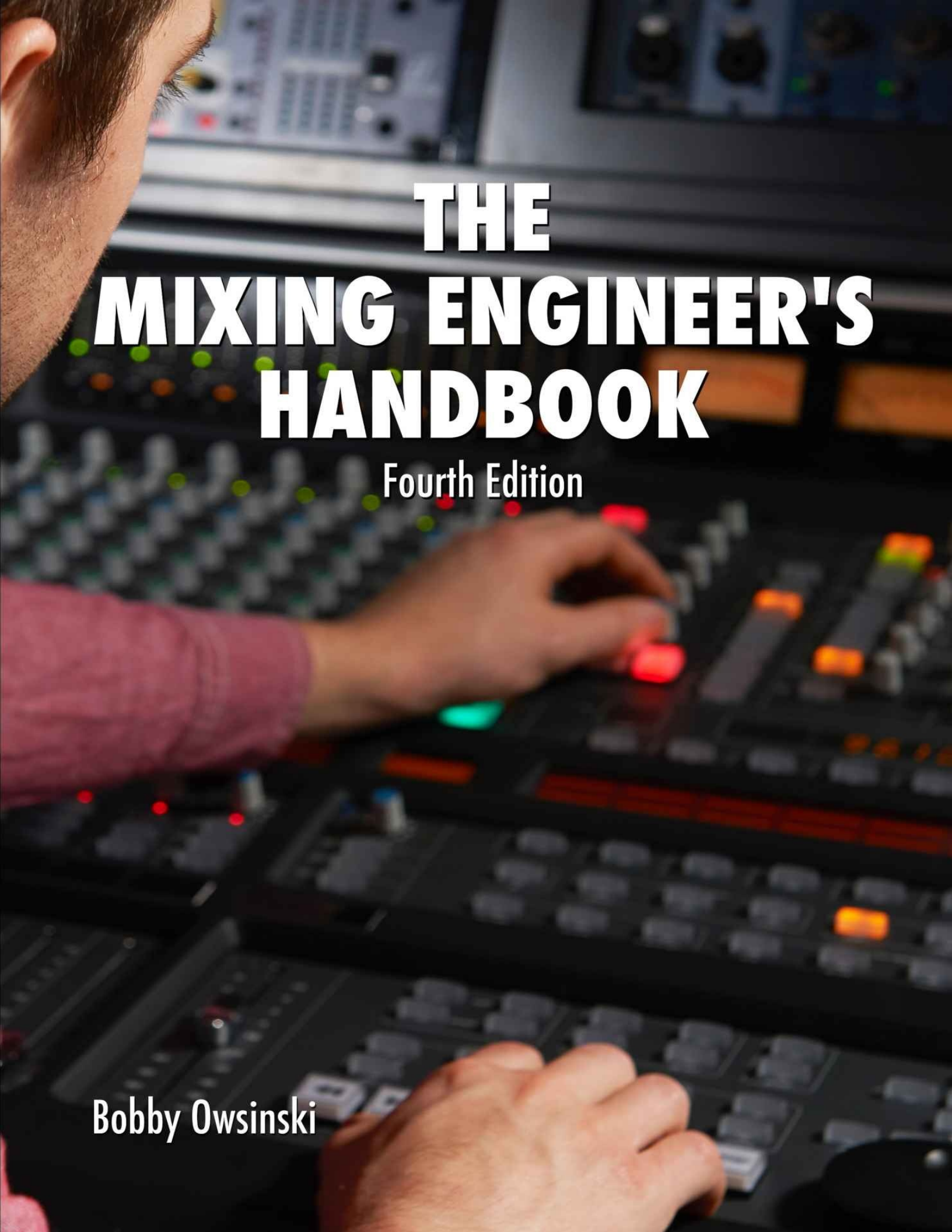

THE MIXING ENGINEER'S HANDBOOK

Fourth Edition

Bobby Owsinski

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The Mixing Engineer's Handbook 4th edition
by Bobby Owsinski

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Introduction

Welcome to the fourth edition of *The Mixing Engineer's Handbook*. In the four years since I wrote the second edition and the 18 years since I wrote the original book, the recording industry has truly undergone a huge paradigm shift. Recording budgets have decreased significantly, the number of major studio facilities has dropped to just a handful in each major media center, and the rise of the digital audio workstation has made it possible for just about anyone to record at home for a minimal investment.

All the more reason to update this book. Mixing techniques have evolved and adapted to the digital world, and with fewer studios, there are also fewer mentors to learn from.
That said, the classic mixing techniques are more useful than ever, since the basics of balance, equalization, compression, and effects never go out of style.

My main goal in writing this book has always been to preserve these techniques before they're lost to rumor or twisted into irrelevance. Where once these skills were handed down from engineer to assistant, that whole master-apprentice information exchange has almost faded into oblivion, which is all the more reason to have a single repository of techniques.

For the fourth edition, I've added a number of new sections and interviews, updated the interviews from the previous editions, and generally adapted the remaining material so that what's contained herein is much more relevant to today's DAW-based mixing. Since the majority of readers will be working at home in their personal studio, I've put a special emphasis on how the pros use their DAWs, as well as how they adapt their large-console techniques to the home studio.

Just so you know, the reason why I originally wrote the first edition of this book is probably the same reason why you're reading it: to get better at what I do. I noticed that my mixes were somewhat hit or miss. Sometimes they were great, sometimes just okay, and sometimes just plain off the mark. I also noticed that much of the time my mixes didn't have the big-time sound that I heard on the radio. I wanted this sound badly, and the only way I knew how to get it consistently was to ask questions of the engineers who already knew the secret.

While doing research for this book, I found that a common factor among most great mixers was that they usually all had at least one mentor as a result of coming up through the studio ranks. Most great mixers started as assistants, learned by watching and listening to the greats they helped, and had taken a little from all of them as a result.

I didn't do that, however. Being a musician first and foremost, I learned to engineer thanks to my early interests in electronics, which came from wanting to know how the electrons got from my guitar to the speakers of my amplifier. As I became familiar with the recording studio, I was lucky to be offered all sorts of varied session work, from recording jingles to big band to jazz to R&B

to hard rock, but since I never wanted to give up being a musician (which I knew I'd have to do), I never took a proper studio job as an assistant to really learn the trade at the hands of the masters. As a result, my recording skills were always pretty good, but my mixing skills were lacking.

I soon realized that there were many others like me who were good but not great, not because they weren't capable, but because they didn't have the opportunity or access to the methods of the masters. After all, how often does a George Massenburg or Bruce Swedien record in Lincoln, Peoria, Santa Fe, or even smaller towns like Minersville, Millersburg, or Avondale? And unfortunately, because there are fewer real commercial studios left, there's even less of a chance of that happening today than ever before. Not only that, the vast majority of musicians (who inevitably end up as engineers in some capacity) operate in their personal studio anyway.

So the first edition of the book started out very selfishly, as it was meant specifically to meet my needs, but it ended up for you as well. I hope you will benefit from it as much as I have.

And yes, my mixes have gotten much, much better.

Meet The Mixers

When I wrote the first edition of *The Mixing Engineer's Handbook*, my intention was to interview as many great engineers as I could in order to accumulate their various methods and anecdotes simply as background material. The more I got into it, though, the more it became obvious that these interviews were living and breathing on their own and they really should be included in the text; otherwise, a lot of really useful information would be left out. In other words, let them tell you what they do in their own words.

These interviews are contained in Part II of the book. Many of the mixers interviewed in the previous three editions have been re-interviewed, since their mixing methods have changed along with the industry changes.

Every one of the mixers I interviewed for this book was extremely forthcoming, answering just about any question and offering explicit information as to why and how he works. Professional jealousy just does not exist in this industry, at least in my experience, as the general attitude is, "I'll tell you anything you want to know, since no one can do it like me anyway."

As a matter of fact, here's a list of the engineers who contributed to this book, along with some of their credits. I've tried to include someone to represent every genre of modern music (punk to classic to alternative to jazz to classical to R&B to EDM to Latin to rap to orchestral to country), so there's something for everyone. I'll be quoting them from time to time, so I wanted to introduce them early on so you have some idea of their background when they pop up.

Just remember, whenever a "mixer" or "engineer" is referred to in this book, I don't mean your average, run-of-the-mill engineer (hardworking and well meaning as he is). I mean someone who's made the hits that you've listened to and loved. This book is about how these glorious few think, how they work, how they listen, and why they do the things they do. And even though we can't hear as they hear, perhaps we can hear through their words.

Bob Brockman: "Bassy" Bob has been a fixture on the New York studio scene with a wide range of awards and credits that include Mary J. Blige, Toni Braxton, Notorious B.I.G., Babyface, Aretha Franklin, Al Green, the O'Jays, Brian McKnight, Jodeci, Faith Hill, Korn, Laurie Anderson, Vanessa Williams, Christina Aguilera, P. Diddy, Herbie Hancock, the Fugees, Santana, and Sting.

Bob Bullock: Since he moved to Nashville in 1984, Bob has been one of the town's top engineers, trusted by the likes of Kenny Chesney, Shania Twain, George Strait, Reba McEntire, Hank Williams Jr., and Jimmy Buffett, among with many others. Prior to that he saw a different side of the music world while working in Los Angeles with acts such as the Tubes, Art Garfunkel,

Seals and Crofts, Chick Corea, and REO Speedwagon.

Joe Chiccarelli: With credits such as the White Stripes, Alanis Morissette, the Strokes, Jason Mraz, Tori Amos, Etta James, Beck, U2, Elton John, Oingo Boingo, the Shins, Frank Zappa, the Killers, Brian Setzer, and many more, chances are you've heard Joe's work more times than you know.

Richard Chycki: Richard has worked with everyone from Rush to Dream Theater to Aerosmith to Mick Jagger and more. In fact, he has 26 gold and platinum records from Rush alone! Later in the book we're going to discuss one of the largest and most complex music projects ever.

Lee DeCarlo: From his days as chief engineer at New York's Record Plant in the heady 1970s, Lee has put his definitive stamp on hit records that include works by Aerosmith, John Lennon, Kenny Loggins, Black Sabbath, Rancid, and Zakk Wylde, among many others.

Jimmy Douglass: One of the few engineers who can cross genres with both total ease and credibility, Jimmy has done records for artists as varied as Snoop Dogg, Jay-Z, the Roots, Ludacris, Justin Timberlake, Timbaland, Missy Elliott, Otis Redding, The Rolling Stones, Foreigner, Hall & Oates, Roxy Music, and Rob Thomas.

Benny Faccone: Benny is unique in that he's a Canadian from Montreal, but 99 percent of the projects he works on are Spanish. From Luis Miguel to Ricky Martin to the Latin rock supergroup Mana, to Spanish remixes for Boys II Men, Tony Braxton, and Sting, Benny's 14-time Grammy-winning work is heard far and wide around the Latin world.

Jerry Finn: With credits from Green Day to Rancid to the Goo Goo Dolls to Beck, Jerry represented one of the new generation of mixers who knows all the rules but are perfectly willing to break them. Unfortunately, Jerry passed away in 2008, but his techniques and wisdom live on.

Jon Gass: Jon has long been the go-to mixer for a who's who of music superstars, including Madonna, Whitney Houston, Janet Jackson, Celine Dion, Mariah Carey, Mary J. Blige, Usher, Babyface, Earth, Wind & Fire, Lionel Richie, John Mellencamp, and many more.

Don Hahn: When it comes to recording and mixing a 45- to 100-piece orchestra, there's no one better than Don, with an unbelievable list of credits that range from major television series to such legends as Count Basie, Barbra Streisand, Chet Atkins, Frank Sinatra, Herb Alpert, Woody Herman, Dionne Warwick, and a host of others (actually, 10 pages more).

Andy Johns: Andy Johns needs no introduction because we've been listening to the music that he's been involved with for most of our lives. With credits such as Led Zeppelin, Free, Traffic, Blind Faith, The Rolling Stones, and Van Halen (to name just a few), Andy has set a standard that most mixers are still trying to live up to.

Bernie Kirsh: From virtually all of Chick Corea's recordings to Quincy Jones, Stanley Clarke, Joe DeFrancesco, and Al Di Meola, Bernie has certainly made his mark as one of the top engineers in the world of jazz.

Nathaniel Kunkel: One of the most in-demand mixers in the business, with credits that range from James Taylor, Lionel Richie, and Sting to Good Charlotte, Fuel, and Insane Clown Posse, Nathaniel represents the best of the next generation of mixers.

George Massenburg: From designing the industry's most heralded audio tools to engineering classics by Little Feat, Earth, Wind & Fire, Dixie Chicks, James Taylor, Billy Joel, Lyle Lovett, and Linda Ronstadt (to name only a few), George needs no introduction to anyone even remotely connected to the music or audio business.

Greg Penny: Born into a music-business family to bandleader/producer Hank Penny and hit recording artist Sue Thompson, Surround Music Award winner Greg Penny seemed destined for a life in the studio. Indeed Greg's production aspirations resulted in hits with k.d. lang, Cher, and Paul Young among others, but a meeting with Elton John while in his teens turned into an award-winning mixing journey with the legend many years down the road.

David Pensado: Over the last two decades, Dave has taken mixing to a new level in artistry, having mixed megahits for superstars such as Christina Aguilera, Justin Timberlake, Kelly Clarkson, Pink, Black Eyed Peas, Beyonce, Shakira, and Michael Jackson, among many others. Well known in the business way before his popular Pensado's Place web video series, Dave not only is on the cutting edge of technology, but has thought long and hard about the more cerebral aspects of mixing as well.

Elliot Scheiner: With a shelf full of industry awards (seven Grammys, an Emmy, four Surround Music Awards, the Surround Pioneer and Tech Awards Hall of Fame, and too many total award nominations to count) from his work with The Eagles, Steely Dan, Fleetwood Mac, Sting, John Fogerty, Van Morrison, Toto, Queen, Faith Hill, Lenny Kravitz, Natalie Cole, Beyonce, the Doobie Brothers, Aerosmith, Phil Collins, Aretha Franklin, Barbra Streisand, and many, many others, Elliot has long been widely recognized for his artful and pristine mixes.

Andrew Scheps: Andrew Scheps has brought a perfect combination of old- and new-school skills to his work with a who's who of superstar artists, including Red Hot Chili Peppers, Metallica, U2, Justin Timberlake, Jay-Z, The Rolling Stones, Linkin Park, Jewel, Neil Diamond, and Adele.

Ken Scott: Legendary producer/engineer Ken Scott began his career working with The Beatles on The White Album and Magical Mystery Tour; on six David Bowie records, including the seminal Ziggy Stardust album; and with Pink Floyd, Elton John, Duran Duran, Jeff Beck,

Supertramp, Procol Harum, Devo, Kansas, Mahavishnu Orchestra, and many more. To put it mildly, he's an absolute icon in the recording industry, having been a part of records that have conservatively sold more than 200 million units.

Ed Seay: Ed has become one of the most respected engineers in Nashville since moving there in 1984, helping to mold hits for major hit-makers such as Blake Shelton, Lee Brice, Martina McBride, Ricky Skaggs, Dolly Parton, Pam Tillis, Highway 101, Collin Raye, and a host of others.

Allen Sides: Although well known as the owner of the premier Ocean Way Studio complex in Los Angeles, Allen is one of the most respected engineers in the business, with credits that include Josh Groban, Michael Jackson, Chris Botti, Barry Manilow, Neil Diamond, Mary J. Blige, and Faith Hill, as well as many major film scores.

Don Smith: With credits that read like a who's who of rock and roll, Don has lent his unique expertise to projects by The Rolling Stones, Tom Petty, U2, Stevie Nicks, Bob Dylan, Talking Heads, The Eurythmics, The Traveling Wilburys, Roy Orbison, and Iggy Pop, among many more. Don is another who unfortunately passed away way too soon, but hopefully this book will keep his unique techniques alive.

Ed Stasium: Ed has made some great guitar albums like the ones by The Ramones, The Smithereens, and Living Color, but he has also worked with the likes of Mick Jagger, Talking Heads, Soul Asylum, Motorhead, and even Gladys Knight and the Pips and Ben Vereen.

Bruce Swedien: Maybe the most revered of all engineers, Bruce has a credit list that could take up a chapter of this book alone. Although the biggest Michael Jackson albums (Off the Wall, Thriller, Bad, Dangerous) would be enough for most mixer's resumes, Bruce can also include such legends as Count Basie, Tommy Dorsey, Duke Ellington, Woody Herman, Oscar Peterson, Nat "King" Cole, George Benson, Mick Jagger, Paul McCartney, Patti Austin, Edgar Winter, and Jackie Wilson, among many, many others.

For those of you who don't have the time or desire to read each interview, I've summarized many of their working methods in Part I of the book.

Please Note: Just because you read this book doesn't automatically guarantee that you'll become a platinum-selling mixer who makes lots of money and works with big-name recording artists. You'll get many tips, techniques, and tricks from the book, but you still need ears and experience, which only you can provide. All this book can do is point you in the right direction and help a little on the way!

Also keep in mind that just because one best-selling mixer might do things a certain way, that doesn't mean that's the only way to do it. In fact, you'll notice that what works for one may be

completely opposite of what works for another, yet they both produce great mixes. You should always feel free to experiment, because, after all, whatever works for you is in fact the right way.

PART I

MIXING

TECHNIQUES

Chapter 1

Some Background

Before we get into the actual mechanics of mixing, it’s important to have some perspective on how this engineering skill has developed over the years.

The Evolution of Mixing

It's obvious to just about everyone who's been around long enough that mixing has changed over the decades, but the why's and how's aren't quite so obvious. In the early days of recording in the '50s, mixing was minimal at best because the recording was made with a single-track mono tape machine and a big recording date meant that all four of the studio's microphones were used. Of course, over the years recording evolved from capturing an unaltered musical event to one that was artificially created through overdubs, thanks to the innovation of Sel-Sync (the ability to play back from the tape machine's record head so everything stayed in sync), introduced in 1955. The availability of more and more tracks from a tape machine lead to larger and larger consoles, which lead to computer automation and parameter recall that became a required feature just to manage the complexity of these far larger sessions. With all that came not only an inevitable change in the philosophy of mixing, but even a change in the way that a mixer listened or thought as well.

According to the revered engineer/producer Eddie Kramer (engineer for Jimi Hendrix, Led Zeppelin, KISS, and many superstars), "Everything was 4-track [when I started recording], so we approached recording from a much different perspective than people do nowadays.

My training in England was fortunately with some of the greatest engineers of the day who were basically classically trained in the sense that they could go out and record a symphony orchestra and then come back to the studio and do a jazz or pop session, which is exactly what we used to do. When I was training under Bob Auger, who was the senior engineer at Pye Studios, he and I used to go out and do classical albums with a 3-track Ampex machine and three Neumann U47s and a single three-channel mixer. With that sort of training and technique under my belt, approaching a rock-n-roll session was approaching it from a classical engineering standpoint by making the sound of a rock band bigger and better than it was. But the fact of the matter was that we had very few tools at our disposal except EQ, compression, and tape delay. That was it."

English mixer Andy Johns, who apprenticed under Kramer and eventually went on to equally impressive credits with the Rolling Stones, Led Zeppelin, Traffic, Van Halen, and others, goes a step further. "You know why [The Beatles seminal album] *Sgt. Pepper* sounds so good? You know why [The Jimi Hendrix Experience] *Are You Experienced* sounds so good, almost better than what we can do now?

Because when you were doing the 4 to 4 [mixing down from one four-track machine to another to open up additional tracks for recording], you mixed as you went along. There was a mix on two tracks of the second 4-track machine, and you filled up the open tracks and did the same thing again. Listen to 'We Love You' by The Stones. Listen to 'Hole in My Shoe' by Traffic. You mixed as you went along; therefore, after you got the sounds that would fit with each other, all you had to do was adjust the melodies.

Nowadays, because you have this luxury of the computer and virtually as many tracks as you want, you don't think that way anymore."

And indeed, once more tracks were available and things began to be recorded in stereo (and now in multichannel surround), the emphasis has turned from the bass anchoring the record to the big beat of the drums as the main focal point. This is partially because typical drum miking went from just overhead and kick-drum mics to the now-common occurrence of a mic on every drum, since the consoles now incorporated more microphone inputs and there were plenty of tracks to record on.

Since the drums could be spread out over 6 or 8 or even 20 tracks, they could now be listened to more carefully during the mix, since they didn't have to be premixed along with the bass onto only one or two tracks. Instead of the drums being thought of as just another instrument equal to the bass, now they demanded more attention because more tracks were used.

At that point (approximately 1975), thanks to the widespread use of the then-standard 24-track tape machine, mixing changed forever, and, for better or for worse, it began to evolve into what it is today.

With more tracks came more mix complexity. Mixing suddenly became a multi-man operation, with four or five sets of hands on different console parameters, every one being given a job to execute during the mix. Mixing became a communal performance by the engineer, producer, assistant, and band members. Of course, it was impossible to execute the mix perfectly all the way through with that much human interaction, so mixes became a collection of sections edited together, but they were holistic and organic and part of the charm of the late '60s- and '70s-era records.

The demand for more mixing precision brought about console automation, first affecting only the console faders and mutes. Now it was possible to reduce the number of humans involved with a mix, since only the console parameters such as EQ and effects sends required manual dexterity. Soon the demands for remixes from record-label executives required that the massive gear setups had to be rebuilt in order to update mixes. The mixing performance had to be recreated even though all that was required was maybe only the vocal level be raised by a dB. This brought about the need for "Total Recall," the feature that made SSL consoles a must-have for every major studio.

While it was now possible to manually recall every position of every parameter on a console, assistants still had to fill in elaborate sheets to manually recall every piece of outboard gear as well. A typical recall setup could take three or four hours alone before the first notes of the song even came out of the speakers.

The next innovation was “resettable” consoles that not only would remember all of your parameter settings, but would automatically reset them so the assistant didn’t have to manually do it. As consoles became larger and larger to the point where 56 inputs was soon considered small, this was almost a necessity. Of course, the outboard gear still had to be reconnected and reset by hand, and with some mixers using 20, 30, or even more pieces during a large mix, life in the studio became more and more complex.

But an interesting turning point occurred around 2001. With the computer-based digital audio workstation now becoming more and more the centerpiece of the studio, much of the automation and effects began to take place inside the DAW (“in the box” became the commonly used phrase), eliminating the need for much of the outboard gear used on every mix. Soon mixers became more comfortable with the sound of mixing completely inside the DAW, and thanks to digital controllers such as the Avid Icon that supplied faders and knobs, they had the same tactile experience as in the analog world of consoles.

Mixing in the box had another big effect on the music business, though. With album-project budgets dropping to the point where they almost matched the price of buying a full DAW setup, many mixers were suddenly faced with the scenario of “We can either pay for you or for the studio, but not both.” This forced many top mixers to move their base of operations from a commercial studio into a studio inside their homes.

Today even the power of free or low-cost DAW applications goes far beyond what major acts were used to using from the ’50s through the ’80s. It’s an amazing time to be an engineer, if you have a handle on what the tools at your disposal are able to accomplish.

Different Mixing Styles

Once upon a time, engineers worked for one particular studio, and one of the reasons a client would book time there was to get the services of that particular engineer. Because the engineer was tied to a specific region of the world, a unique mixing style for the area developed (much like what happened with the music), thanks to engineers and clients exchanging tips and tricks with one another. As a result, up until the late ’80s or so, it was easy to tell where a record was made just by the sound of the mix.

Today there’s less of a distinction than there used to be between the mixing styles of different areas. There’s been a homogenization of styles in recent years mostly because engineers now mix in a variety of locations around the world, and many have relocated to new areas, transplanting their mixing styles along the way.

That said, the mixing styles of today can be traced to four major styles from the past, where most mixes fell into one of them: New York, Los Angeles, London, and Nashville. If you listen to records from the ’80s and ’90s, you can distinctly hear these styles.

The New York Style

The New York style used to be perhaps the easiest to identify because it featured a lot of compression, which makes the mix very punchy and aggressive (just like New Yorkers). In many cases, the compressed instruments (mostly the rhythm section) are even recompressed several times along the way. It seems that every New York engineer that I've ever talked to (even the transplanted ones) used the same trick, which is to send the drums (sometimes with the bass) into a couple of busses, send that through some compressors, squeeze to taste, then add back a judicious amount of this compressed rhythm section to the mix through a couple of channels.

This effect can be enhanced even further by boosting the high and low frequencies (lots of boost in many cases) to the compressed signal as well. (More on this "New York Compression Trick" later in the book in Chapter 9, The Dynamics Element: Compression, Limiting, Gating, and De-Essing.)

The LA Style

The LA style exhibited a somewhat more natural sound, which, although compressed, is done to a less obvious degree than the New York style. There's also less effects layering than the London style, but a good bit of delayed reverb is added. The LA style has always tried to capture a musical event and then sonically augment it, rather than recreate it. Some good examples would be any of the Doobie Brothers or Van Halen hits of the '70s and '80s.

The London Style

The London sound was a highly layered musical event that borrowed from the New York style in that it would be pretty compressed but had multiple effects layers that put each instrument into its own distinct sonic environment. Although musical arrangement is important to any good mix, it's even more so a distinctive characteristic of a London mix.

What this means is that many mix elements appear at different times during a mix, some for effect and some to change the dynamics of the song. Each new element would be in its own environment and, as a result, would have a different ambient perspective. A perfect example of this would be Hugh Padgham's work with the Police, or just about anything produced by Trevor Horn, such as Seal or Grace Jones or Yes's "Owner of a Lonely Heart."

The Nashville Style

Nashville has gone through various phases through the years where the mixing style has evolved. At one point in time, the mixes were so dependent on the artist that the vocal sat way out in front of the music bed, sometimes almost to the point where they both seemed almost disconnected.

The Nashville style today has evolved (some might say devolved) from what it was during the ’60s and ’70s to become much more like the modern compressed version of the LA style of the ’70s. Says engineer/producer Ed Seay:

> *“Back when I used to listen to my dad’s old Ray Price and Jim Reeves country records, they weren’t very far from what pop was in the early ’60s: very mellow, big vocals, very subdued band, very little drums, strings, horns, lush. Mix-wise, there wasn’t really too much difference in an Andy Williams record and one of the old Jim Reeves records.*
>
> *What happened was that country got too soft-sounding. You’d cut your track and then do some sweetening with some horns and strings. At one time strings were on all the country records, and then it kind of transformed into where it’s at today, with almost no strings on country records except for big ballads. For the most part, horns are completely dead. They’re almost taboo. Basically it’s rhythm track–driven and not really very far off from where pop was in the mid to later ’70s. The Ronstadt “It’s So Easy to Fall in Love” and “You’re No Good,” where you hear guitar, bass, drums, keyboards, a slide or steel, and then a vocal background; that’s pretty much the format now, although fiddle is used also. Ironically enough, a lot of those guys that were making those records have moved here because, at this point, this is one of the last bastions of live recording.”*

Nowadays there’s far less difference between styles than there was during the ’50s to the ‘80s, but variations still do exist. Although the style differences blur on most music, electronic dance music still has considerable variation divided around the traditional geographic boundaries of New York, LA, and London, with additional pockets in major cities around the world.

Other Styles

Increased globalization has had its effect on regional styles. Where once upon a time Philadelphia, Memphis, Ohio, Miami, and San Francisco all had sub-styles of the Big Four, the globetrotting lifestyle of most A-list engineers in the ’90s caused a homogenization of these regional styles. Where at one time most studios had house engineers, the market became predominately made up of freelancers that frequently traveled from studio to studio and project to project, bouncing between different cities (and therefore styles) as easily as flipping the channel on a TV. Where at one time, an engineer might change studios but remain located in a specific area all his working life, it became commonplace for an engineer to relocate to several major media centers during the course of his career. Because of this movement, a cross-pollination of styles started to blur the distinction between the Big Four in the ‘90s.

Today the differences are far fewer than they used to be. Now everyone uses pretty much the same gear, which wasn’t true in the heyday of analog. During those years (which really started to wane in about 2001–2002), a studio in each city had a different gear list, from consoles to monitors to tape machines to outboard gear. As a result, everyone had to work a little differently, so the style (as well as the city’s musical environment) was different as a result. Not so anymore. As a result, the distinction between mixing styles will be less and less as the years go on.

Where the difference will remain is in the philosophy that’s handed down from the A-team mixers of each city to their assistants. Because there are fewer real commercial studios these days, there are fewer assistants learning a particular style as a result. Thanks to the Internet and books like this, the styles have become more or less the same.

12 Reasons Why Studio Mixing Is Different From Live Mixing

You may have a lot of experience mixing a live band, but mixing in the studio is a distinctively different experience. The thought process is different, the mindset is different, the approach is different, and the chain of command is different.

In an effort to contrast these two different experiences, let's move from the most evident differences to those that are, shall we say, a bit more subtle.

1. **Repertoire.** Most live gigs rarely change repertoire much from gig to gig. You can hone the mix for each song the more times you gig. In the studio, each song is unique and fresh, and when it's finished, it's on to the next one.

2. **Scrutiny.** On a live gig, your mix is gone as soon as the song is over. In the studio, what you do is under a microscope and will likely be analyzed, dissected, and reorganized, all in the name of making the mix stronger.

3. **Equipment.** The gear you use on a live gig won't always translate to the studio. You choose the gear for a gig based upon versatility, durability, and general ruggedness. The only thing that counts in the studio is the sound. While one size might fit all on a gig in terms of gear such as compressors, delays, and reverbs, the presets that are frequently used usually make for a boring studio mix, and don't forget that the studio world is now one of plugins. The studio requires a wide range of sonic possibilities, so you'll need to have a number of gear or plug-in choices to get there.

4. **Leadership.** On a gig you have a bandleader that makes the set list, counts off the songs, may direct the solos, and ends the songs, but how you mix is pretty much up to you. In the studio you're answering to a hierarchy consisting of the producer, the artist, and most likely, someone at the record label if the artist is signed. The producer is the final decision-maker, with ultimate authority over everything you do, although the artist has much say in the final product as well.

5. **Nuance.** The little things count in the studio. Everything you do can be critical to a mix, so nuances are just as important as the basic balance of the instruments (as you'll see all throughout this book). During a live gig, the nuances are usually gone in the wind, overcome by the stage volume, the acoustics, and the attention span of the players and audience. In the studio, everything you do is scrutinized because it's all captured. What that means is you've got to be great every song.

6. **Etiquette.** You might get away with being a jerk on a live gig since the band or others on

the crew usually will put up with you (to a point) as long as you do your job well. Not so in the studio. If you make someone feel even slightly uncomfortable for any reason, chances are you probably won't be asked to do another project with them.

7. **It's hard work.** That's not to say that mixing a four- or five-hour gig isn't difficult, but there's the variation of mixing a whole set list of different songs plus the glory of the audience feedback. In the studio, the only feed- back you get is from the producer and artist, and 99 percent of the time they're analyzing how you can make the mix better rather than singing your praises. And the level of concentration is definitely up a few notches. You can sometimes breeze through a gig, almost losing yourself in your mixing. In the studio, every moment of every track counts and requires your utmost attention.

8. **Preparation.** Live gigs sometimes may not even require a rehearsal to learn the songs. The studio mixer requires both system and personal preparation before even a single fader is raised, as you'll see in Chapter 3, "Mix Preparation."

9. **Approach.** Live mixers strive to get the same sound every gig, while a studio mixer strives to achieve a different sound on every song. Studio mixing requires experimentation and skill in working with constantly changing sounds and sonic characters, which is quite the opposite from a live mixer.

10. **Pace.** On a gig there's a constant pace: show up, set up, soundcheck (maybe), gig, and tear down. In the studio the pace is usually set by a budget and/or a deadline. You may only have a limited amount of time to finish the mix, make any tweaks, and deliver it, regardless of whether the mix feels finished or not.

11. **The required skill set.** For live mixing, the skill set requires that you know how to mix in an ever-changing acoustic environment and have a basic instrument/vocal balance technique. The studio requires your hearing to be more nuanced with a different reference point as to what sounds good or bad and how it will translate to other speakers outside the studio, plus you need a greater knowledge of what the gear and plugins are capable of.

12. **The live wolf pack and the studio lone wolf.** Most live performances require a group and a sizable supporting cast (unless you're working with a DJ or a solo singer/songwriter) that mostly stays the same every gig. Studio mixers are independent and usually work with different people on every project, or mix completely by themselves with little interaction with the client whatsoever.

Learning How to Mix

Mixing can't be taught; it has to be learned. Being a good mixer is the sum total of all of your mixing experiences. It's multiple "aha" moments that eventually get strung together. It's the sharing of experiences with your friends, clients, and books like this. It's the pat on the back and a "job well done" from the artist you tried so hard to please. It's the comparison of your mix with one you admire, only to realize that you're not as far away as you once were. In other words, it's the experience of doing it over and over again, each time experimenting until your ears sharpen and you develop a technique that you're personally comfortable with.

Becoming a good or even great mixer takes practice. There are a number of tips and tricks that you'll see in both this book and the Lynda.com *Audio Mixing Bootcamp* (go to lynda.com/trial/bowsinski for a free 10 day trial) that I authored that can help accelerate your learning curve, but you still have to put in the time.

That said, a lot of different factors enter into a mix. The mix depends heavily upon the song, the musical genre, the musicians, the quality of the recording, and the arrangement—things that you may or may not have under your control.

There will be times when a well played, arranged, and recorded song will make the mix go so quickly and easily that you feel like a genius. Then there will be others when 100 tracks of poorly conceived and executed audio mush will bring you to your knees.

Both situations, as well as everything in between, are valuable. This is how mixing is learned, by experience.

If you're mixing mostly your own songs, make an effort to mix some of your friends' work as well, even as a trial, because the experience will be a powerful learning tool. Likewise, try to do mixes in different musical genres from what you normally work in. Different sounds and arrangements will open up your ears.

Above all, keep on mixing. It's the best way to learn!

Chapter 2

Monitoring

A mixer is only as good as the monitors he uses, which is why this significant piece of hardware is examined before any specific mixing technique. If the monitors don't work with the environment or if the mixer doesn't interact well with them, then all the other tips and techniques may not help you as much as you'd like.

> *"I can use almost any studio as long as the monitors are good, because then at least you have the confidence in what you're hearing. It doesn't matter whether you're using crappy mics; if you're getting a sound that you like coming off the monitors, then you know it works."*
>
> *—Ken Scott*

The Listening Environment

Probably the single most important area that gets overlooked in most home studios is the listening environment. While it's possible that you can get lucky with a balanced, even sound by just setting up a couple of nearfield monitors in your room without thinking much about it, usually that's not the case. The average garage, living room, or bedroom was never intended as a listening space and has little in the way of acoustic treatment as a result.

Acoustically treating your room to smooth out some of the frequency imbalances can be cheaper and easier than you think, but that subject is beyond the scope of this book. You can read more on how to do that in my *The Studio Builder's Handbook* (Alfred Publishing). That said, here are a number of very simple things you can do to help your acoustic environment that consist of nothing more than where you place your monitors in the room.

Overcoming Potential Acoustic Problems

Here are a few things to avoid if you can help it. These may seem like small fixes, but in some cases they can make a dramatic difference in what you hear.

1. **Avoid placing speakers up against a wall.** This usually results in some strong peaks in the low-frequency response. The farther away you can get from the wall, the less it influences the frequency response of your monitors, so the smoother that response can be.

2. **Avoid the corners of the room.** Worse than the wall is a corner, since it will reinforce the low end even more than when placed against a wall. Even worse than that is if only one speaker is in the corner, which will cause the response of your system to be lopsided.

3. **Avoid being closer to one wall of the room than the other.** If one speaker is closer to a side wall than the other, once again you'll get a totally different frequency response between the two because the sonic reflections from each wall is different. It's best to set up directly in the center of the room if possible.

4. **Avoid different types of wall absorption.** If one side of the room contains a window and the other is painted drywall or something like carpet or acoustic foam, once again you'll have an unbalanced stereo image because one side is brighter than the other. Try to make the walls on each side of the speakers the same in terms of absorption quality.

Monitors: Which Ones?

Which monitor speaker is best for you? There are certainly plenty of choices, and currently there's no single favorite among the great mixers. Probably as close to a standard as we've ever had was the Yamaha NS-10M, closely followed by the Auratone 5C (Figure 2.1). Because NS-10s are no longer made, they're becoming less and less of a standard. Auratones have fallen out of favor since their peak of popularity during the '70s, but many mixers still use them as an additional reference (although sometimes only one in mono), especially now that they're being manufactured again.

Figure 2.1: An Auratone 5C

That said, you don't necessarily need a set of monitors that are considered a standard or are even popular among mixers. It's possible to get great mixes out of virtually any set of speakers in just about any room, and that includes headphones as well. The trick is that you have to have enough listening time to get a reference point as to what sounds good or bad when you play your mix back elsewhere. That's why mixers began to take their own speakers with them wherever they went (or asked for NS-10s) in the first place. It was a sound that they were familiar with, and since they were nearfields, the room didn't come too much into play during the mix, so they could be more comfortable with the result.

One of the good things about today's monitors is that since most of them now come with their own on-board amplifier, you no longer have to worry about buying an external amp to power them. That eliminates one of the major heartaches of choosing a monitor, since the match with the amplifier is critical, and the speaker could sound very different when paired with a different amp. With the on-board amp now perfectly matched to the monitor, this is no longer a problem. Plus,

the overall monitor package is less expensive as well.

That said, here are a few tips to consider before choosing a monitor.

1. Don't choose a monitor because someone else is using it. Monitors are a lot like guitars. Just because Slash uses a Les Paul doesn't mean that it's also right for you. It might not match the music you make, or it might be too heavy for your particular body frame. Same with a monitor. Just because your favorite mixer uses a set of Dynaudio BM5As, that doesn't mean they'll be right for you, too. They may not fit the type of music you work on or your room, or they may not even appeal to the way you hear.

2. Listen to the monitors before you buy them. Before a pro purchases a monitor, he'll take his time to listen to them under a wide range of conditions. Why shouldn't you do the same? It's understood that this may be a problem if you don't live near a big city with a pro audio dealer. Even if you do, you may not have a relationship with one that allows you a personal demo in your own environment, but that shouldn't stop you from listening to them first. This is a serious purchase, so don't take it lightly. Take the trip to your local pro audio or music store and prepare to spend some time listening. Listen to everything and spend as much time with each model as you can. What should you listen for? Here's how to evaluate a monitor:

- **Listen for even frequency balance.** While listening to a piece of music that you know well, check to see whether any frequencies are exaggerated or attenuated. This is especially important in the midrange crossover area (usually about 1.5k to 2.5 kHz). Listen especially to cymbals on the high end, vocals and guitars for the midrange, and bass and kick drum on the low end.

- **Make sure the frequency balance stays the same at any level.** The less the frequency response changes as the level changes (especially when playing softly), the better. In other words, the speaker should have roughly the same frequency balance when the level is quiet as when it's loud.

- **Make sure you have enough output level without distortion.** Be sure that there's enough clean level for your needs. Many powered monitors have built-in limiters that stop the speaker or amplifier from distorting but also may keep the system from getting as loud as you might need it to be.

- **Above all, don't buy a set of speakers without listening to them.** It's usually very difficult for them to live up to your expectations if you've not heard them first. In fact, it's not a good idea to buy any speaker unless you're really in love with it. You'll have to listen to these monitors for a lot of hours, so you might as well like what you hear.

3. Listen with source material that you know very well. The only way to judge a monitor is to

listen to material that you're very familiar with and have heard in a lot of different environments. This gives you the necessary reference point to adequately judge what you're listening to. If you don't have anything that you've recorded yourself that you know inside and out, use a favorite CD that you consider to be well recorded.

Remember, don't use MP3s here! Use only CDs or a playback system with an even higher quality 24-bit source, such as files on a flash drive or a personal digital recorder. That should give you some idea of the frequency response of the system.

One thing I learned when writing speaker reviews for EQ magazine over the course of five years is that you can easily get used to just about any speaker if you use it enough and learn its strengths and weaknesses. It also helps to have a solid sonic reference point that you're sure of to compare the sound with. For instance, if you know how things sound in your car, then adjust your mixes so they work when you play them there. Believe it or not, that's still a go- to place for many major mixers to reference their work.

> *"It still amazes me that with all these great and expensive tools, I can still throw it in the car and in like two seconds I immediately know what's wrong."*
>
> *—Ed Seay*

Basic Monitor Setup

Too often musicians and engineers haphazardly set up their monitors, and this is a leading cause for mix problems later on down the line. How the monitors are placed can make an enormous difference in the frequency balance and stereo field and should be addressed before you get into any serious listening. Here are a few things to experiment with before you settle on the exact placement.

Check the Distance Between The Monitors

If the monitors are too close together, the stereo field will be smeared with no clear spatial definition. If the monitors are too far apart, the focal point or "sweet spot" will be too far behind you, and you'll hear the left or the right side individually but not both together. A rule of thumb is that the speakers should be as far apart as the distance from the listening position. That is, if you're 4 feet away from the monitors, then start by moving them 4 feet apart so that you make an equilateral triangle between you and the two monitors (Figure 2.2). A simple tape measure will work fine to get it close. You can adjust them either in or out from there.

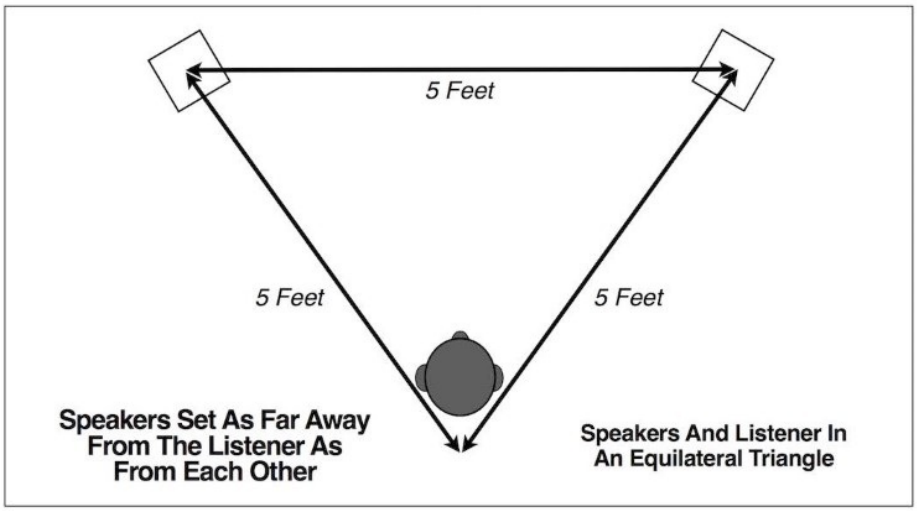

Figure 2.2: Set the monitors in an equilateral triangle
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That being said, it's been found that 67 1⁄2 inches from tweeter to tweeter at the distance of a console meter bridge seems to be an optimum distance between speakers and focuses the speakers just behind your head (which is exactly what you want).

A real quick setup that we used to use in the days of consoles is to open your arms as wide as possible to each side and place the monitors at the tips of the fingers of each hand. This seemed to work well because of the built-in depth that the console meter bridge would provide, but it doesn't really apply in these days of workstations where the monitors are a lot closer to you. If that's the case, go back to the equilateral triangle outlined above.

Check The Angle Of The Monitors

Improper angling will also cause smearing of the stereo field, which could mean that you'll have a lack of instrument definition as a result. The correct angle is determined strictly by taste, with some mixers preferring the monitors to be angled directly at their mixing position and others preferring the focal point (the point where the sound from the tweeters converges) anywhere from a foot to about three feet behind them to eliminate some of the "hype" of the speakers.

> ***TIP**: A great trick for getting excellent left/right imaging is to mount a mirror over each tweeter and adjust speakers so that you can see your face clearly in both mirrors at the same time when you are in your mixing position.*

Check How The Monitors Are Mounted

Monitors that are mounted directly on top of a desk or console meter bridge without any decoupling are subject to comb-filter effects, especially in the low end. That is, the sound from the monitor causes the desk or console to resonate, causing both the desk and speaker to interact as certain frequencies are either cancelled or reinforced. This is what's known as phase cancellation, and it causes a subtle yet very real blurring of the sound.

As a result, it will be just a little harder to hear your low end distinctly, which makes it more difficult to EQ. Phase cancellation can be more or less severe depending on whether the speakers are mounted directly on the desk or metal meter bridge or mounted on a piece of carpet or similar material (which is very popular).

One of the quickest ways to improve the sound of your monitor system is to decouple your speakers from whatever they're sitting on. This can be done with a commercial product such as Primacoustic's Recoil Stabilizers (see Figure 2.3), or you can make something similar relatively cheaply with some open-cell (closed-cell will work, too) neoprene or even some mouse pads. As mentioned earlier, carpet is sometimes used as well, although it's not nearly as effective as neoprene.

Figure 2.3: Equator D5 on isolation pad
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Decoupling the subwoofers that you're using from the floor can really help, too. While sometimes the coupling with the floor can make your low end feel bigger, it will be a lot clearer and more

distinct if decoupled. Auralex even has a product for this called the SubDude HD (see Figure 2.4), although you can probably put together a DIY setup that can work just as well.

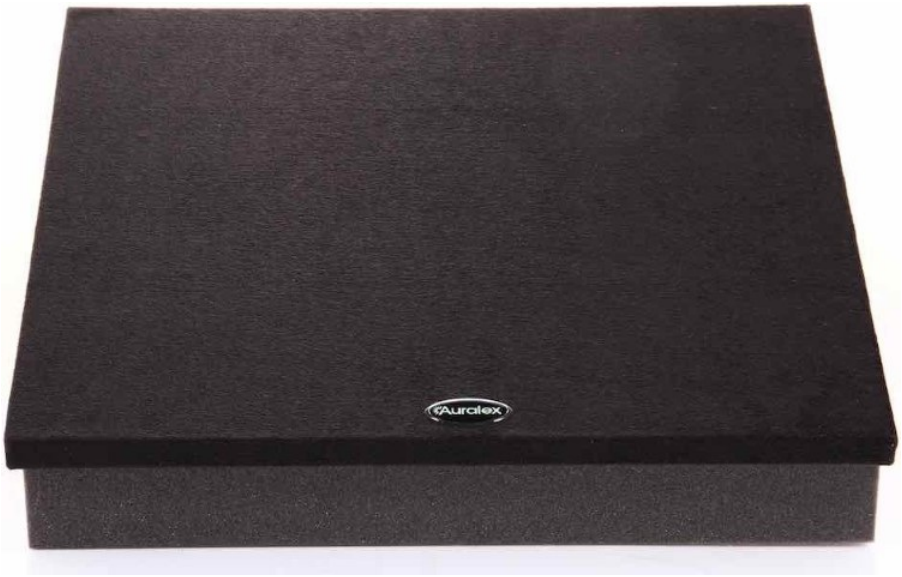

Figure 2.4: Auralex Subdude HD with subwoofer mounted on top
© Primeacoustic, All Rights Reserved

Regardless of the brand, model, and type of speakers that you use, decoupling is a cheap and easy way to improve your sound right away.

> ***TIP**: The best solution is to mount your monitors on stands just directly behind the desk or meter bridge. Not only will this improve the low-frequency decoupling, but it also can greatly decrease the unwanted reflections off the desk or console.*

Check How The Monitor Parameters Are Set

Most powered monitors have anywhere from a single volume control to a wide array of parameter selections. Regardless of how many parameters your monitors might have, be sure that these controls are set correctly for the application and are the same on each (see Figure 2.5).

Figure 2.5: Monitor speaker parameter controls.

© Yamaha Corporation of America, All Rights Reserved

Check The Position Of The Tweeters

Many monitors are meant to be used in an upright position, yet users frequently will lay them down on their sides. This results in a variety of acoustic anomalies that deteriorate the sound. If the monitors are laid on their sides, most mixers prefer that the tweeters of a two- or three-way speaker system be on the outside, thereby widening the stereo field. Occasionally, tweeters to the inside works, but usually it results in a smearing of the stereo image. Experiment with both, however, because you never know exactly what will work until you try it (see Figure 2.6).

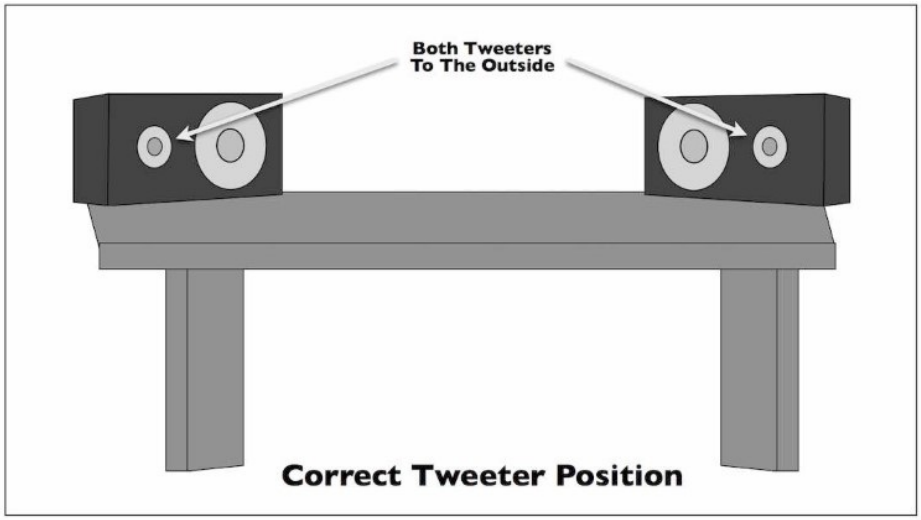

Figure 2.6: Tweeter position

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Check The Desk Or Console Itself

If you're using a console, its angle; the type of materials used for the panels, knobs, and switches; the type of paint; and the size and composition of the armrest all make a difference in the sound due to reflections causing phase cancellation. If the sound of the monitors on top of the meter bridge is unacceptable, then try moving them toward you with extenders or put them on stands behind the console (don't forget to decouple them).

Three Steps To Adding A Subwoofer

It's not unusual for musicians and engineers who are doing a lot of work in their home studio on bookshelf-sized speakers to crave more bottom end. As a result, the first thing they think about is adding a subwoofer to their monitor system. That's all well and good, but there are a few steps that you can follow that might help make your venture into low-frequency territory a lot easier.

A. **Do you really need a subwoofer?** Before you make that purchase, it's a good idea to be sure that a sub is actually necessary. Here are a couple of things to check out first:

- **Are you monitoring at a loud enough level?** This is a trap that people with home studios fall into; they don't listen loudly enough, at least for a short period of time. First of all, if you listen too quietly, your ears begin to emphasize the mid-frequencies. This is great for balance but bad for judging the low end of a song. Crank up your monitors to a moderately loud level, at least when you're working on the low-frequency end of the spectrum. If you still don't have enough low end, go on to the next point.

- **Do you have an acoustic problem in your room?** Chances are that either your monitors are too close to the wall or they're placed at a point of the room length where standing waves cause some of the low end to cancel out. This is more likely to be the cause of just one area of the low-frequency spectrum rather than the entire low end, though. Just to be safe, move your speakers a foot or so backward and forward and see whether you get some of the low end back. If not, move on to…

B. Purchase a subwoofer from the same manufacturer as your main monitors. The easiest way to get a smooth-sounding low end that doesn't cause you more grief than it's worth is to buy a sub to match the monitors that you use most of the time. That means if you're using JBLs, choose a JBL sub that's made specifically for that system; if you're using Genelecs, do the same, KRKs, the same, and so on. This will make a huge difference, especially at the crossover-frequency point where the mains cross over to the sub. It's usually extremely difficult to get that area to sound natural if you mix brands.

C. Calibrate your sub correctly. Most musicians and engineers who choose to use a sub just randomly dial in the level. You might get lucky and get it right, but it's more likely that your level will be off, causing a number of unbalanced-sounding mixes until you finally figure it out. Here's how to calibrate the sub to your system:

1. Without the sub connected, send pink noise to your main monitors. At the listening position and while listening to one monitor only, use an SPL meter (just about any of them

will do to get you in the ballpark, even an iPhone app) and adjust the level of the monitor until it reads 85 dB. The SPL meter should be set on C Weight and Slow. Repeat on the other channel and set that so it also reads 85 dB.

2. Turn off the main monitors. Send pink noise just to the subwoofer. Set the level of the SPL meter so it reads 79 dB. Although it may seem like it will be lower in level, 79 dB works because there are fewer bands of low frequencies than high (three for the low and eight for the high), so this number takes that into account. You might have to tweak the level up or down a dB, but this will get you into the ballpark.

3. If there's a polarity switch on the sub, try both positions and see which one has the most bass or sounds the smoothest in the crossover area. That's the one to select.

If you follow these steps, you'll find that integrating a subwoofer into your system (if you decide you need one) will be as painless as possible.

Mixing On Headphones

Sometimes it's just not possible to listen to your monitors to mix. When it's late at night and your kids, significant other, or neighbor is in the next room, separated only by paper-thin walls, you have no choice but to try to mix on headphones. Mixing on headphones does have four significant downsides, though:

- You can't wear them for as long as you need to (8, 10, 12 hours) before your head and ears get tired from the extra weight.
- You have a tendency to turn them up, which can lead to some quick ear fatigue, again limiting your ability to mix for long periods.
- Because most of the more expensive professional headphones really sound great, you get a false sense of what the mix is like (especially on the low end), and it causes you not to work as hard at getting the frequency balance of the mix right.
- The vast majority of the audience won't listen on real headphones after the mix is completed. Because a mixer is always aiming for a mix that sounds great on a wide variety of speakers on which the material is played, you want to stay in that realm if possible and even listen on some crappy speakers as a check if possible.

Headphones (not to be confused with common earbuds) just sound too good for that.

That said, headphones do have their place. They're great for editing because they allow you to hear clicks, pops, and inconsistencies that you may otherwise miss while listening on speakers, and they're a great check for panning and reverb tails when mixing. That doesn't mean that you should use them for an entire mix, but if you have no choice, then by all means go for it. Just make sure that you listen to some other material that sounds great on your speakers first so that you have a reference point of what sounds good and what doesn't.

How Loud (Or Soft) Should I Listen?

One of the greatest misconceptions about music mixers (especially the great ones) is that they mix at high volume levels all the time. While some do, many mixers also find that they get better balances that translate well to the real listening world by monitoring at conversation level (79 dB SPL) or lower, although most will go up to a much higher level briefly when checking the low end of a mix.

Using high levels for long periods of time is generally not recommended for the following reasons:

- First the obvious one: Exposure to high volume levels for long periods of time may cause long-term physical damage to your hearing.
- High volume levels for long periods of time will cause the onset of not only ear fatigue, but general physical fatigue as well. This means that you might effectively only be able to work 6 hours instead of the normal 8 (or 10 or 12) that's possible with lower levels.
- The ear has different frequency-response curves at high volume levels that overcompensate on both the high and low frequencies. This means that your high-volume mix will generally sound pretty limp when played at quieter levels.
- Balances tend to blur at higher levels, so what works at high volume won't necessarily sound great when played at a quieter level. However, balances that are made at quieter levels always work when played louder.

Now this isn't to say that all mixing should be done at the same level and it should all be done quietly. In fact, music mixers (as opposed to film, which always has a constant SPL level) tend to work at a variety of levels—up loud for a minute to check the low end, moderate while checking the EQ and effects—but the final balances are frequently done quietly.

> *"Generally speaking, when I put up the mix, I'll put it up at a fairly good level, maybe 105 [dB SPL] and set all my track levels and get it punchy and fun-sounding. And if I listen loud, it's only for very short periods of time. It's rare that I would ever play a track from beginning to end loud. I might listen to 20 seconds or 30 seconds of it here and there, but when I'm actually down to really detailing the balance, I'll monitor at a very modest level. I would say at a level that we could have a conversation and you could hear every word I said."*
>
> *—Allen Sides*

“I mix at different levels. I try not to mix too loud because it’ll wear you down and fool your perspective. Sometimes it’s very valuable to turn things down, but there’s an up and down side to both. If you listen too soft, you’ll add too much bass. If you listen too loud, you’ll turn the lead vocals down too much.”

—Ed Seay

“I’ll monitor way loud to see what rocks. I’ll monitor at a nominal level to get sounds together, then I’ll monitor about 5 dB over background noise to bring all the elements into focus. If a mix works at 30 dB SPL, 25 dB SPL, it’ll almost always work a lot louder. If you can hear everything at that low a level, then when you turn it up you’ll have a very even balance. That’s the way to get everything in the same plane, by listening extremely low.”

—George Massenburg

“I listen quietly as much as I can. It’s hard to check kick-drum level when it’s quiet, so certainly you have to push it up every once in a while, but I fatigue pretty quickly when listening at loud levels. I can make better emotional and timbre decisions before I fatigue.”

—Nathaniel Kunkel

Listening Techniques

Most experienced mixers have determined that regardless of how familiar they are with their monitors, they need some additional assurance that what they're hearing is really what they're hearing. As a result, a variety of listening techniques have been developed over the years that many still use.

Listening On Multiple Monitors

The number of monitor references that are used is an important aspect to getting the balance of a mix just right. Although a mixer may do most of his work on a single system, it's common to check the mix on at least two (maybe more) other sources as well. This might be the main soffit-mounted monitors if mixing in a commercial studio, the nearfield monitors of choice, and an alternative, which could be Auratones, NS-10s, computer speakers, or just about anything else. Couple that with a boombox, car stereo, or stereo in the lounge, and the average of all these systems should tell you what you need to know about the mix. There are even some plugins that simulate all these different listening environments like Audified's *MixChecker*, which is worth checking out.

Most mixers will settle on a set of monitors they feel they can trust, learn its strengths and weaknesses, and then do a check on a smaller set. This can be anything from a small computer extension speaker, to earbuds (more and more popular), to even the speaker in a laptop.

The alternate speaker is used simply as a balance check to make sure that one of the instruments isn't either too loud or too soft in the mix. Also, one of the arts of mix balance is getting the kick drum and bass guitar to speak well on a small system, which is why an alternative monitor system is so important.

The second set of monitors doesn't have to be great. In fact, the worse they are, the better. Even a set of $10 computer speakers will do. The idea is to have a second set that will give you an idea of what things sound like in that world, since unfortunately, there are a lot more people listening on lousy monitors than good ones these days.

Other mixers prefer to listen outside of the main listening area with the door open.

> *"What I'll do about an hour before printing the mix is prop open the control-room door and walk down the hall or into the lounge where the music has to wind its way out the door. It's real valuable to see if you hear all the parts, and it's real easy to be objective when you're not staring at the speakers and looking at the meters."*
>
> *—Ed Seay*

> *"I'm a big one for hallway. I like listening around the corner or on a blaster."*
>
> *—George Massenburg*

> *"I'll walk out of the control room and listen to it right outside the door. It's interesting to hear what it sounds like through the crack in the door.*

Things pop out."

—Joe Chiccarelli

> ***TIP**: Good television-commercial producers use a similar technique: Flip to a single mono Auratone (or similar small speaker), lower the volume to just perceptible, and see whether it still sounds like a record. Then raise the volume a tiny bit, walk out into the hall, and see whether you still like it.*

Of course, the car still seems to be the gold standard of engineers, producers, and artists alike. Once upon a time it was important to hear what it sounded like in that environment in the event that the song received radio play. Over time, the car stereo has evolved into a system that most people are very familiar with, considering the time each of us spends in our cars these days. Regardless, don't underestimate the power of the car for checking a mix.

Listening In Mono

Sooner or later your mix will be played back in mono somewhere along the line, so it's best to check what it will sound like when that happens so you're not surprised later. Listening in mono is a time-tested operation that can actually give the mixer the ability to discern phase coherency, balances, and even panning.

Phase Coherency

When a stereo mix is combined into mono, any elements that are out of phase will drop in level or even completely cancel out. This could be because the left and right outputs are wired out of phase (pin 2 and pin 3 of the XLR connector are reversed), which is the worst-case scenario, or perhaps because an out-of-phase effect causes the lead vocal or solo to disappear. In any event, it's prudent to listen in mono once in a while just to make sure that a mono disaster isn't lurking in the wings.

Balances

Many engineers listen to their mix in mono strictly to balance elements together, since they feel that they hear the balance better this way. Listening in mono is also a great way to tell when an element is masking another.

> *"I listen in mono an awful lot and find it's great for balances. You can easily tell if something's fighting something else."*
>
> *—Joe Chiccarelli*

Panning

Although not many engineers are aware that their stereo panning can be improved while listening in mono, this is in fact a good way to achieve a level of precision not available in stereo.

> *"I check my panning in mono with one speaker, believe it or not. When you pan around in mono, all of a sudden you'll find that it's coming through now and you've found the space for it. If I want to find a place for the hi-hat, for instance, sometimes I'll go to mono and pan it around, and you'll find that all of a sudden it's really present, and that's the spot. When you go to stereo it makes things a lot better."*
>
> *—Don Smith*

Chapter 3

Mix Preparation

Preparing for the mix can be as critical as the mix itself, since it allows for a more comfortable and efficient mixing session that minimizes mistakes and hassles. This prep occurs before the first fader is raised but sets the stage for an easier and mistake-free mix by having the files properly labeled and all the assignments, effects, and routing preset beforehand.

We can break the session setup into two distinct tasks: prepping your session and prepping yourself. Let’s look at each.

Prepping Your Session

Prepping your mix has evolved over the years. Once upon a time it consisted of labeling the console, setting up the outboard gear, and biasing the tape machine. Today it’s more about labeling the files and arranging the session layout inside your DAW. Let’s take a closer look.

Make a Session File Copy

It's not a good idea to work on the original session file that you're given, because if you have to go back to the beginning because the file becomes corrupted or you just need to begin again with a clean slate, the original file will be altered. The first thing to do is make a copy of the session file, then name it something descriptive like "songtitle mix" so it's easy to locate (see Figure 3.1). This keeps your previous session file safe if you ever have to go back to it.

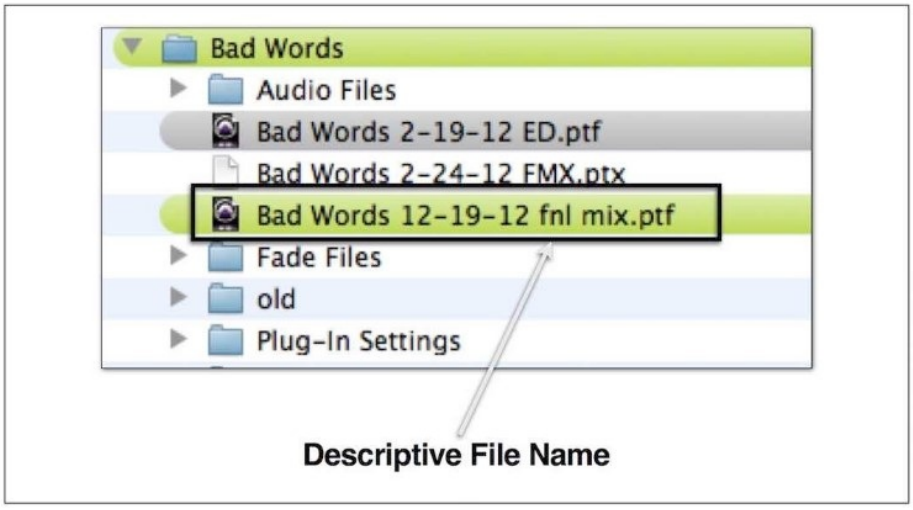

Figure 3.1: A descriptive file title
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Many mixers put a date in the file name, but that's not necessary the time is almost always built into the metadata and can be easily determined by looking at the file info. It's not uncommon to have multiple versions of the same session during the same day; one way to differentiate one from another is to use letters of the alphabet or numbers at the end of the title, such as "Love Park mix 9-9-12a," "Love Park mix3," and so on.

If your DAW allows it, color-coding the file also makes it easier to identify. One way is to start with a preplanned series of colors that show the stage of completion, such as red for the beginning of a mix and green for when it's finished, or vice versa. Of course, use whatever colors work for you.

> ***TIP**: It's also a good idea to make a copy of the session file on another hard drive, flash drive, online backup, or anyplace that you can easily grab it if for some reason your work file becomes corrupted.*

Tweak The Track Timing

No matter how accomplished the players on a recording session are, many times there's some portion of a player's recording that doesn't feel quite right. While recording, however, you normally have enough time to have the musician play his part until it's perfect or punch in all the suspect parts as you go along.

Usually, the timing of the basic tracks will be tweaked right after your basic tracking session so you have a solid rhythm section to overdub against, but if you haven't done that or you're just now discovering some sections that don't feel right (which happens a lot), prepare for the joys of slipping and sliding time. You can read about some track timing techniques in the "Adjust the Timing" section of Chapter 11, "Advanced Techniques."

Check The Fades

Check each fade-in to be sure that nothing is getting cut off too soon. Check each fade-out on elements such as background vocals or doubled instruments to be sure that the releases are the same. (You can read more about doing this in Chapter 11.)

Eliminate Noises

Now is the time to clean up each individual track. Although the noises might not sound too bad when played against all the other tracks, after everything is mixed and mastered you'd be surprised by how something that was once buried can now come to the forefront and bother you. Also, by eliminating any extraneous noises, all the tracks magically sound more distinct and uncluttered.

- **Trim the heads and tails.** Trim all the extra record time at the beginning and end of each track, regardless of whether it was recorded during basics or overdubs. Add a fade-in and fade-out to eliminate any edit noise (see Figure 3.2).

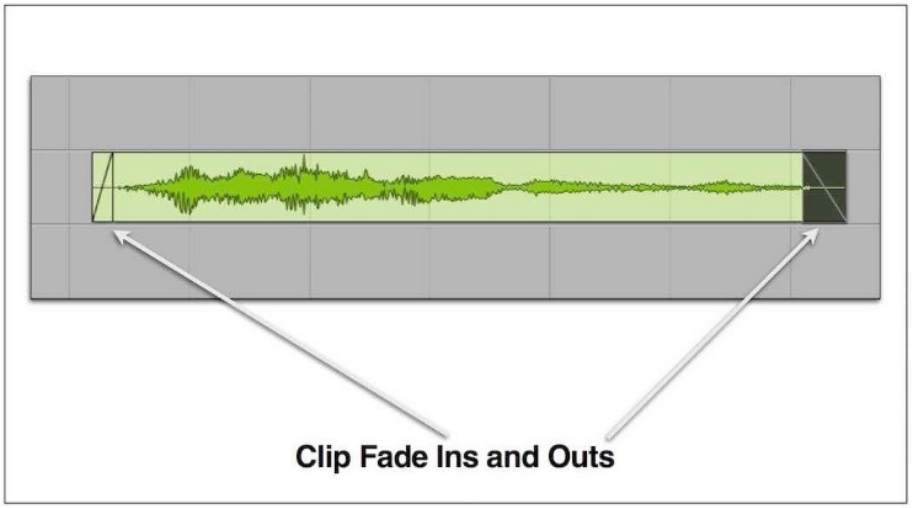

Figure 3.2: Fade-ins and fade-outs on a trimmed clip
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- **Crossfade your edits.** One of the biggest problems for A-list mixers is when they get a session that's full of edits is that the edits click and pop because they don't contain any crossfades. Even if you can't hear a click or pop, it's a good practice to have a short crossfade on every edit to eliminate the possibility of an unwanted noise (see Figure 3.3).

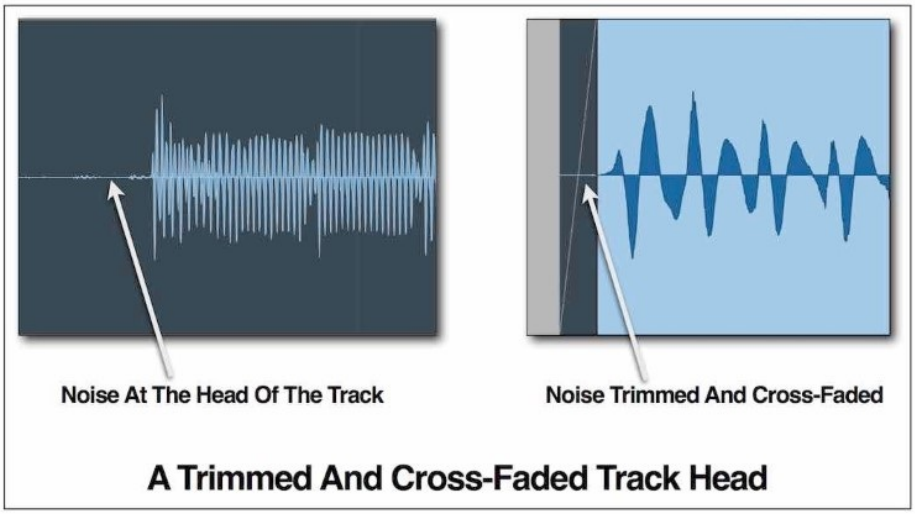

Figure 3.3: A trimmed and crossfaded track head
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- **Delete extra notes from MIDI tracks.** Delete any extra "split" notes that were mistakenly played. You might not hear them when all the instruments are playing, but just like the noise at the beginning of tracks, they have a tendency to come to the forefront after things get compressed.

> *"Today there's a lot of cleanup stuff that's sort of expected as part of the mixer's job, which should be a production thing. Production has gotten a lot lazier in the last 5 or 10 years. Now it's unbelievable how much garbage, extraneous stuff, clicks and pops, and unlabeled tracks that you get even in major projects."*
>
> *—Bob Brockman*

Comp Your Tracks

Comping shouldn’t be left for mixing, as it’s something that’s normally taken care of directly after an overdub session where the vocal, guitar, or anything else is recorded with multiple takes. That being said, if you still have some vocal or overdub comping to do, now’s the time. You can read more about comping techniques in Chapter 11, “Advanced Techniques.”

Tune Your Tracks

Inevitably there's always a note that's a bit sour and needs tuning. Whether you use Auto-Tune, Elastic Audio, or any other pitch correction plug-in, make sure that the timing isn't thrown off when the note is shortened or lengthened. You can learn more about correcting pitch in Chapter 11.

Consolidate Your Tracks

Once all of your edits, comping and tuning is complete, it's a good idea to consolidate all the edits of a track into a single solid clip. Not only does this make it easier on the eyes, but it will keep you from accidentally moving or deleting an edited clip during the mix. Where this might not be an advantage is if you're planning on changing the level, effect or plugin just on a single clip during the mix (providing your DAW has these features). That said, most DAWs allow you to backtrack on un-consolidate if needed.

Arrange Your Tracks

These days a typical session has plenty of tracks that won't be used in the final mix. Deleting or hiding these tracks and then putting the rest in a logical order can be the single most useful thing you can do while prepping your mix. Here are a number of steps to take if you'll be mixing the song yourself, or especially if you'll be giving it to someone else to mix.

Delete Empty Tracks

Empty tracks take up space in your edit and mix windows without adding anything useful, so it's best to delete them. During tracking or overdubs, it frequently makes sense to have empty tracks readily available so you can go instantly to another take or overdub with a minimum of time, but if you've gotten to the mix stage without using some of them, they're only taking up space on your desktop, making it more difficult to see some of the channels you really need. Delete them.

Deactivate And Hide Unused Tracks

Any tracks that you know won't be needed (sometimes they're marked or color-coded beforehand) just soak up your computer's system resources. Even with a power machine, these resources may become a precious commodity if you end up using a lot of plugins during the mix. Deactivate the tracks and then hide them from the timeline and mix panels so they don't distract you and take up desktop space.

Reorder Your Tracks

Reordering tracks into logical groups of instruments or vocals makes finding individual instruments or vocals easier during the mix. The idea is to group any similar instruments or vocals together, so all the guitars are next to each other, the drums and percussion are next to one another, and keyboards, horns, strings, and all the vocals are together. This makes it easier not only to find a track, but also to group them later if needed.

Color-Code The Tracks

This isn't absolutely necessary, but it does make things easier to find if your DAW app has this ability. For instance, all the drums might be red, guitars blue, the vocals yellow, and so on.

Correctly Label the Tracks

Many workstation apps automatically assign a name to any new track that has been recorded, but usually the names don't relate to the instrument (see Figure 3.4). It's really easy to mistake one track for another and turn a fader or parameter knob up and up and wonder why nothing is happening, only to find that you're tweaking the wrong track. That's why it's important to clearly label each track that has a name like "gt166" to something descriptive like "guitar" or " rhythm gtr." Consider this to be the same as identifying the channels of a console with console tape, and make it a habit when prepping.

Figure 3.4: Relabeled tracks
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Even if it's your own project, don't assume that you'll be the only one to ever mix it. Label each track so that anyone can easily identify it. Labels such as "John 1" and "Paul 4" don't mean anything to anyone but you. Who's John? What is Paul playing or singing? Mark each track logically.

> *"One issue is organization, because the track labeling is often really poor, and I find I'm spending hours of prep time before I can even get into a mix. I get lots of tracks that are just labeled 'Audio 1,' 'Audio 2,' or 'Bob 1' or 'Bob 2.' I don't care that it's Bob singing. I need to know exactly if it's a high harmony in the chorus."*
>
> *—Joe Chiccarelli*

Make Your Decisions

One of the unfortunate parts about our digital world of unlimited tracks is that it's really easy to put off decisions about which tracks to use until the mix. This can be difficult even if you're familiar with all the tracks, but nearly impossible if you're seeing the session for the first time.

For instance, if you have seven different guitar takes, you don't need to keep them all. Make a decision that will work best and then deactivate and hide the others. If you have four mic setups for each guitar, trust me, you don't need them all. Make a decision up front which one will work and save yourself a lot of time. Chances are that the difference between them isn't as great as you thought it would be.

> *"Make decisions as much as you can along the way. Everyone is too willing to let any decision wait until the last minute instead of making it on the spot. Today you see people piling on track after track and waiting until the mix to sort it out. That makes everything take a lot longer and makes it harder to mix because you're not sure how everything is going to fit together."*
>
> *—Ken Scott*

> *"That's the thing that does take a bit more time these days in that there are so many possibilities and people don't make decisions now because they know they don't have to commit to something until the very end."*
>
> *—Joe Chiccarelli*

Insert Section Markers

Time markers (sometimes called memory locations) are a major timesaver in any DAW and essential to an efficient mix. If not done already, now is the time to mark each section of the song (see Figure 3.5). Most veteran mixers insert a marker a bar or two before each new section so there's some pre-roll, and also make sure that other points in the song, such as drum fills, accents, or even the halfway point in a section, are marked as well.

Figure 3.5: Marker or memory locations

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Create Groups And Subgroups

Subgroups are a way to pre-mix a number of channels before they're sent out to the main outputs of the mixer (see Figure 3.6). This allows you to not only control the level of several channels with a single fader, but also EQ or add effects to all of those channels via the subgroup channel as well.

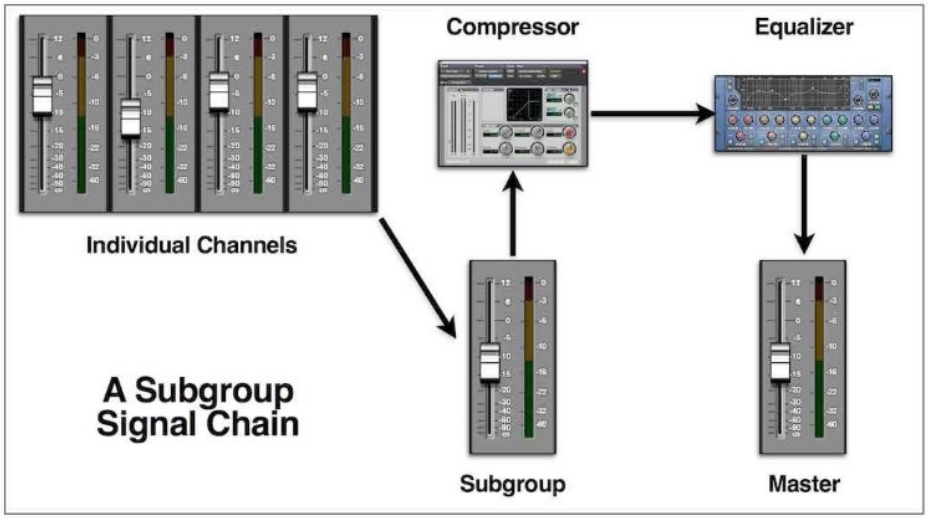

Figure 3.6: Subgroup diagram
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Subgroups can be useful during mixing because they allow you to group similar elements of the mix so you can make adjustment by instrument sections, rather than individually (see Figure 3.7). A mix can go much faster if the subgroups are created and the particular instrument or vocal channels are assigned to them ahead of time.

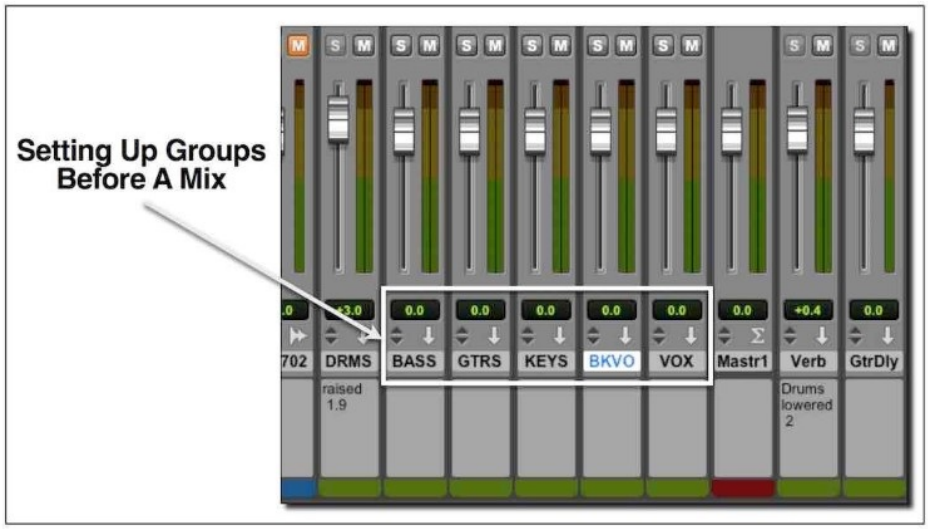

Figure 3.7: Setting up subgroups
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Typical groups might include drums, guitars (if there's more than one or they're in stereo), lead vocals (if there's a double or there's a track for each song section), background vocals, horns, strings and synths, or any instrument or vocal recorded in stereo across two mono channels.

Groups (not to be confused with subgroups) are virtual subgroups in that the assigned faders are linked together so when one is moved, they all move in proportion to their balance, but they retain their individual outputs to the stereo master buss (see Figure 3.8).

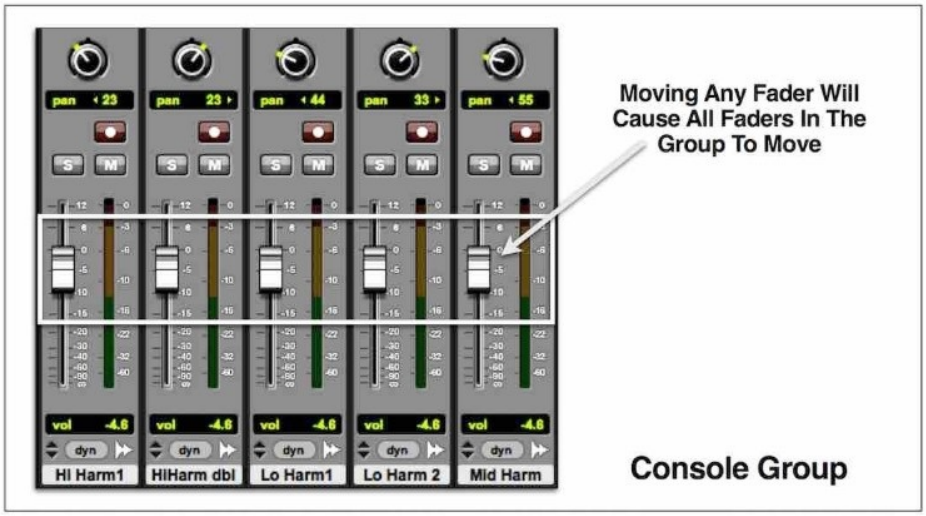

Figure 3.8: Group diagram
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Generally speaking, groups work well with stereo instruments such as piano, organ Leslies, guitars recorded with multiple mics, drum overheads, and the like. Subgroups work best with large groups of instruments such as the drum kit, background vocals, and string sections, although how they're used is purely up to the taste of the mixer. Groups may be used within subgroup channels, as illustrated in Figure 3.9.

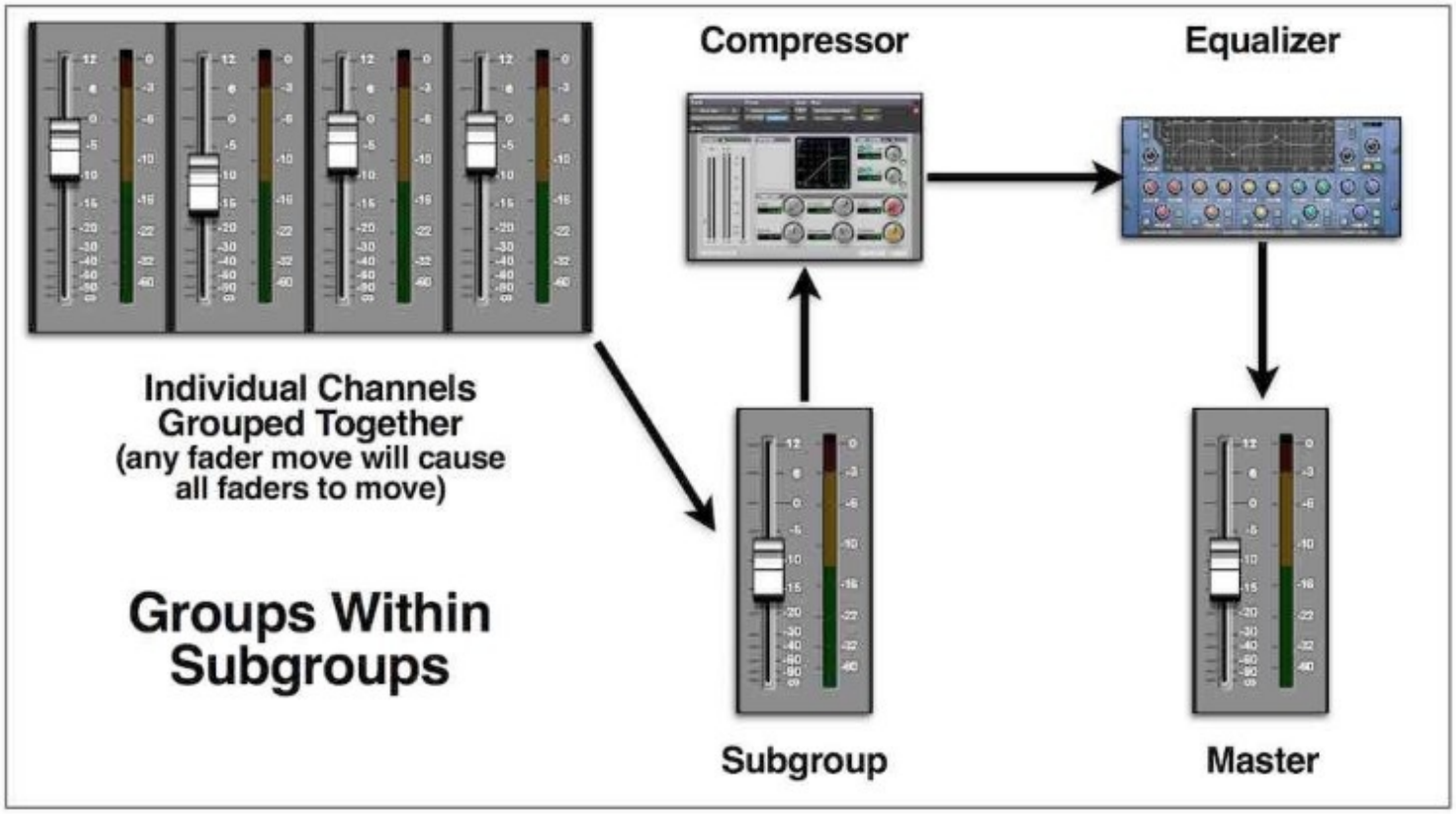

Figure 3.9: Groups used within subgroup channels
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Create Effects Channels

Many mixers have a standard set of go-to effects that they'll set up before they begin a mix, since they know that they'll eventually be put to use at some time during the mix. This will be covered more in Chapter 8, "The Dimension Element," but one quick setup that works well even for tracking, overdubs, and rough mixes utilizes two reverbs and a delay and is set as follows:

- **For drums:** Use a reverb set to a dark room sound with about 1.5 seconds of decay and a pre-delay of 20 milliseconds.
- **For all other instruments:** Use a plate with about 1.8 seconds of decay and a pre-delay of 20 milliseconds. For vocals: Use a delay of about 220 milliseconds with two repeats.

These settings work very well together and create a nice blend of ambience without much tweaking. Another common setup uses two reverbs and two delays, which are set like this:

- **Short reverb:** A room program with the decay set from .5 to 1.5 seconds of decay with a short pre-delay timed to the track. (See Chapter 8 for more detail on how to do that.)
- **Long reverb:** A plate or hall program with a decay set from 1.5 to 4 seconds of decay and a pre-delay of as little as 0 or as much as 150 milliseconds timed to the track (depends on your taste and what's right for the song).
- **Short delay:** A delay of about 50 to 200 milliseconds.
- **Long delay:** A delay from about 200 to 400 milliseconds.

Your own particular starting point might use a lot more effects, or you may prefer to add effects as they're needed during the mix. Regardless, it's a good idea to have at least some effects set up before you begin the mix so you won't break your concentration to set them up later. Some additional effects setups are described later in Chapter 8.

Assign The Channels

Usually you know ahead of time that there are certain tracks that will use a particular effect (such as in the case where the drums or snare will use a short reverb). It's best to assign those channels to the appropriate sends and pan them accordingly before your mix begins, but make sure that the send is set to infinity or off.

> ***TIP**: Randomly assigning busses to groups or sends can get confusing on a large mix. Some mixers choose to group the busses by function. For instance, 1 to 10 for subgroups, 11 to 20 for reverbs, 21 to 30 for delays, and so on.*

Insert Compressors And Limiters

In most modern mixes at least a few channels, such as the kick, snare, bass, and vocal, will usually need a compressor during the mix in order to modify the track's dynamic range. The mixing process can be sped up if the compressors are inserted on those channels ahead of time during mix prep. Remember to leave any compressor or limiter bypassed until you decide to use it later during the mix (see Figure 3.10).

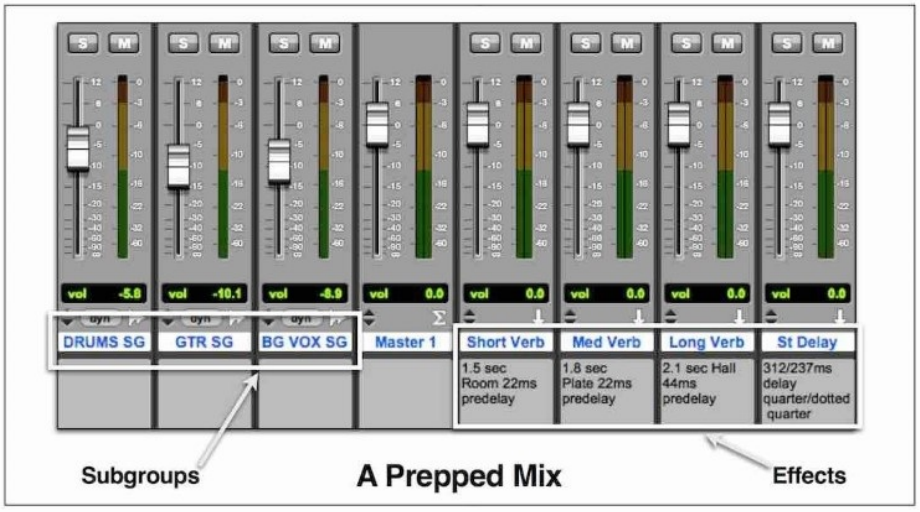

Figure 3.10: A fully prepped mix

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Personal Preparation

One overlooked area of mix prep is personal preparation, which means getting yourself into the best physical and mental state to mix. Each mix requires focus and concentration, and this is where you get yourself into that proper headspace.

Calibrate Your Hearing

Our ears are amazing organs that are capable of hearing sounds so faint that they move the eardrum less than the diameter of a hydrogen molecule. It's important that we first calibrate them to take advantage of their enormous capabilities. Here's how:

- Try to stay in the quietest area that you can for as long as you can before you begin your mix. Concentrate on the sounds that you're hearing and try to identify what they are and the direction they're coming from. Studies have found that this can make your hearing much more acute.
- Stay away from a large meal before you mix, because it will temporarily make your hearing less sharp during digestion.
- To improve your ability to hear faint sounds, relax your jaw or just smile. There are tiny muscles in your jaw that can actually disrupt the action of your eardrums and Eustachian tubes, which control the inner-ear pressure.
- If you will be doing work that requires your attention on a computer monitor screen, even small noises can subtly blur your vision. Turn the level down and try to keep the uninvited noises at bay.

***TIP**: Also remember that closing your eyes while mixing can sometimes improve your hearing by both lessening the distractions and allowing your brain to concentrate additional processing power on that sense.*

> *"I've been mixing with my eyes closed for most of my career. I think I have an easier time visualizing the three- dimensional panorama that's coming out of the speakers. Somehow when I close my eyes it's easier for me to see an instrument or vocal by removing my eyes from the equation altogether."*
>
> *—Bob Brockman*

Get Your Listening Reference Point

Even if you know your room very well, it's still important that you establish your listening reference point before you begin. This is accomplished by playing at least one song or mix that you know well so you have a reference point as to what the room and monitors sound like. Listening to a mix or two will also calibrate your ears to the listening environment, which will help to keep you from over- or under-EQing as you go along.

Prepare For Note-Taking

During the course of a long mix, you'll probably have to take some notes, so have a pen and a pad of paper or even some Post-it notes ready to write on. If you're using a hardware controller, you'll need a roll of console tape. Shurtape FP 726 is the type that can be reapplied to mark the names of the channels without leaving any sticky residue behind.

Make Yourself Comfortable

Most mixes take a while, so both you and your environment need to be comfortable. Make sure your clothes and shoes are comfy, the room temperature is just right, and the lighting is adjusted so you can easily see any monitor screens that you may be using without squinting, because this can lead to fatigue before you realize it. It's also a good idea to have some beverages and a snack ready for later when you need a break.

> ***TIP**: A comfortable chair can provide a greater feeling of well-being, which in turn can heighten your senses, like your most important one for mixing: hearing.*

Take Frequent Breaks

Be sure to take frequent breaks while you're mixing, since it will not only give your ears a chance to rest, but also prolong your ability to work. Some mixers may take a break as frequently as every hour, while others might wait for three or four hours before they feel the need to rest.

A break means a complete break from the listening process, not just from mixing. Getting some air or sunshine, grabbing a beverage, and changing the subject for 10 or 15 minutes goes a long way to a successful mix without you feeling worn out at the end.

Stay Focused On The Mix

The studio isn't a place for playback parties with your friends or potential clients. Every time you stop for as little as a phone call, it breaks your concentration and it's that much harder to get the momentum back. Turn the phones off and check your messages only on breaks. Even then, try to limit return calls to emergencies. Anything that breaks your concentration takes your focus away from the mix.

It's especially difficult to stop a mix to play back a track for your friends, who of course will then play you one that they're working on. Not only is your focus shot at that point, but your ears need to be recalibrated as well. Stay disciplined. Tell your friends that you'll be happy to meet them for a playback session after your mix is completed.

Setting up for a mix is a lot more work than you might have thought, but it's time put to good use. Once these things are out of the way, your files, tracks, mind, and ears are all set for the mix ahead.

> ***TIP:*** *Remember the mantra - "Fix it before you mix it!"*

Chapter 4

The Mechanics Of Mixing

Although most engineers ultimately rely on their intuition when doing a mix, there are certain mixing procedures that they all consciously or unconsciously follow.

Conceptualizing The Mix

By and large, most mixers can hear some version of the final product in their heads before they even begin to mix. Sometimes this is a result of countless rough mixes during the course of a project that gradually become polished, especially now thanks to the ability to recall the previous mix in a digital workstation, if the engineer is mixing a project that he has tracked. Even if an engineer is brought in specifically to mix, many won't even begin until they have an idea of where they're going before they start.

Many engineers who can hear the finished product before they start normally begin a mix the same way. They become familiar with the song either through a previous rough mix or by simply putting up all the faders (virtual or otherwise) and listening for a few passes. Sometimes this is harder than it seems, though. In the case of a complex mix with a lot of tracks, the mix engineer may have to spend some time muting channels and instruments before the song begins to pare down and make sense.

> *"I always try to have a vision of the mix when I start. Rather than just randomly pushing up faders and saying, 'Well, a little of this EQ or effect might be nice,' I like to have a vision as far as where we're going and what's the perspective."*
>
> *—Ed Seay*

> *"If I know the song, then I already have a pretty clear picture of what I'd like it to be. If not, I'll usually get that the first time I listen through a track. It's not so much for the sonics, but more in terms of size, like figuring out how big the chorus will be. Sometimes I'll get really specific ideas about effects that I'll try as well. I think the main thing, especially if it's a song I haven't recorded, is that I go through instrument by instrument to see how it sounds, but what I'm really doing is learning every single part so that when I come to build my balance, I know where everything is going to be."*
>
> *—Andrew Scheps*

> *"Luckily, about half the stuff I do these days I'm producing, where I'm always cutting in mix mode, so it sounds like the record right from the beginning."*
>
> *—Jon Gass*

For better or worse, the engineer's vision might change along the way, thanks to input from the

producer and/or artist. Considering how often a mix is done in a personal studio these days, a major mixer frequently completes the job without the producer and/or artist ever coming to the studio, although many mixers actually prefer the input. However, a vast majority would prefer to start the mix by themselves and have the artist come by to offer suggestions five or six hours later, after the mix begins to take shape.

The Overall Approach

Whether they know it or not (and many mixers aren't aware of how they do it), most great mixers have a method in the way they approach a mix. Although the method can vary depending on the song, the artist, the genre, or even if the mixer tracked the song from scratch or is just coming in for the mix, the technique remains constant.

1. Determine the Direction of the Song

2. Develop the Groove

3. Find the Most Important Element and Emphasize It

The last point may be the most important in creating an outstanding mix. As famed Latin mixer Benny Faccone so succinctly states, "It's almost like a musician who picks up a guitar and tries to play. He may have the chart in front of him, but soon he has to go beyond the notes in order to get creative. Same thing with mixing. It's not just a thing of setting levels any more, but more about trying to get the energy of the song across. Anybody can make the bass or the drums even out."

Tall, Deep, And Wide

Great mixers mix in three dimensions. They think “tall, deep, and wide,” which means that all the audible frequencies are represented, there’s depth to the mix, and it has some stereo dimension as well (see Figure 4.1).

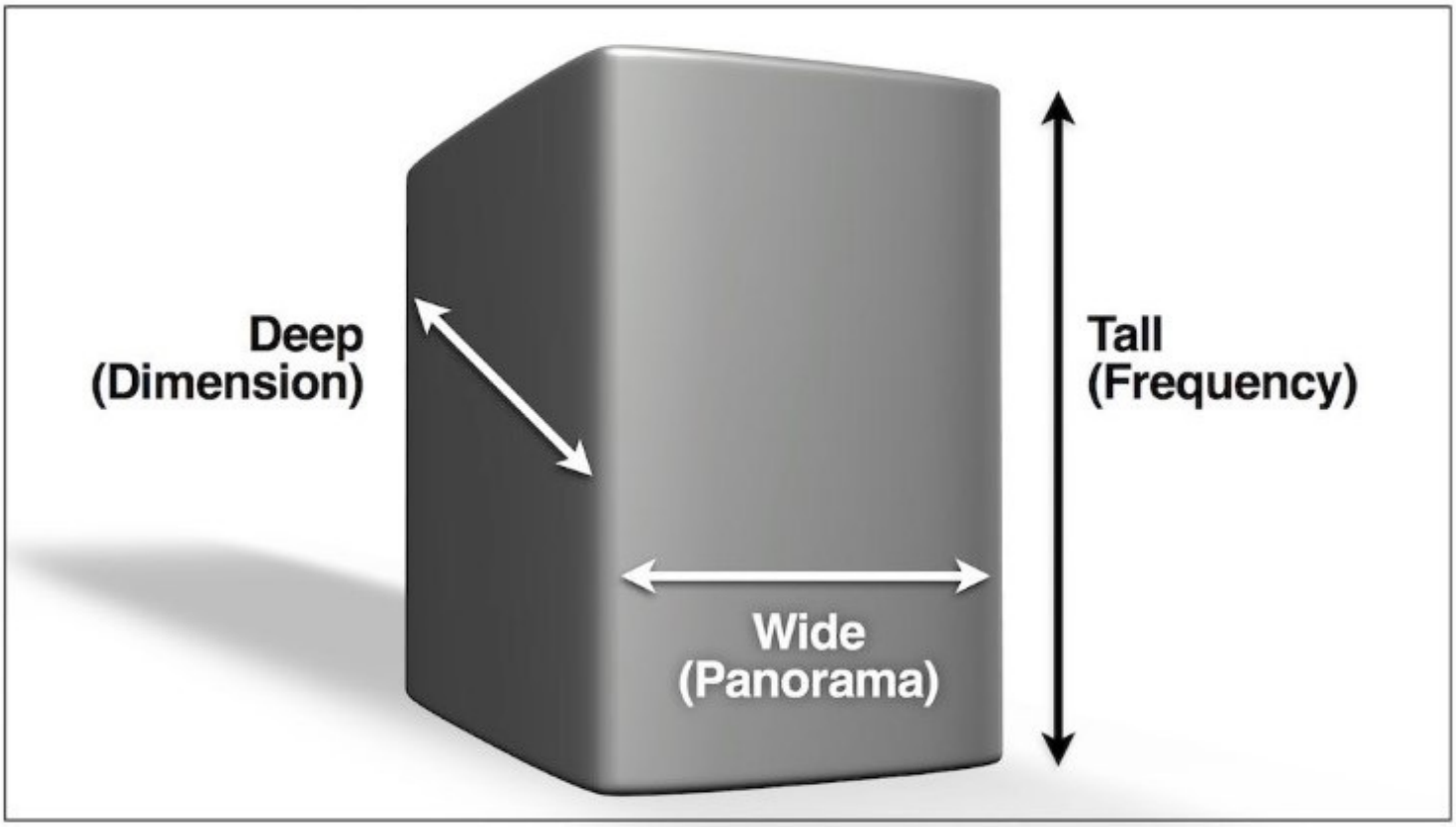

Figure 4.1: Tall, deep, and wide
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The “tall” dimension is the result of knowing what sounds balanced frequency-wise as a result of having a strong reference point. This reference point can come from experience as an assistant engineer listening to what other engineers do, or simply by comparing your mix to some CDs or high-resolution files that you’re very familiar with and appreciate for their fidelity.

Essentially what you’re trying to accomplish is to make sure that all the frequencies are properly represented.

Usually that means that all of the sparkly, tinkly highs and fat, powerful lows are there. Sometimes some mids need to be cut. Most of the time, clarity is what you aim for, but that might not be the case in every genre of music. Again, experience with mix elements that sound good really helps as a reference point.

The effects or “deep” dimension is achieved by introducing new ambience elements into the mix. This is usually done with reverbs and delays (and modulation offshoots like flanging and

chorusing), but room mics, overheads, natural ambience, and even leakage from other instruments play an equally big part in creating the depth element as well.

The panning or “wide” dimension is the process of placing an audio element in the soundfield in such a way that it creates a more interesting soundscape, as well as makes sure that each mix element is heard more clearly.

The Signs Of An Amateur Mix

Before we can talk about how to make a great mix, it's good to be aware of the signs of one that isn't that great. Does your mix have any of these characteristics?

- **The mix has no contrast.** That means that the song has the same musical or sonic texture throughout the entire song, or the mix is at the same level and intensity through the song.
- **The mix has a wandering focal point.** There are holes between lyrics where nothing is brought forward in the mix to hold the listener's attention.
- **The mix is noisy.** Clicks, hums, extraneous noises, count-offs, and sometimes lip-smacks and breaths can be clearly heard.
- **The mix lacks clarity and punch.** The instruments aren't distinct, or the low end is either too weak or too big.
- **The mix sounds distant and devoid of any feeling of intimacy.** The mix sounds distant because of too much reverb or overuse of other effects.
- **The mix has inconsistent levels.** Instrument levels vary from balanced to quiet or too loud, or certain lyrics or instrument lines can't be distinguished.
- **The mix has dull and uninteresting sounds.** Generic, dated, or often-heard sounds are being used. There's a difference between using something because it's hip and new and using it because everyone else is using it.

The Six Elements Of A Mix

Every genre of modern music, be it rock, pop, R&B, hip-hop, country, new age, swing, drum and bass, trance, or any other category featuring a strong backbeat, has six main elements required for a great mix. They are:

- **Balance**. The volume-level relationship between musical elements.
- **Frequency range**. Having all audible frequencies properly represented.
- **Panorama**. Placing a musical element in the soundfield.
- **Dimension**. Adding ambience to a musical element.
- **Dynamics**. Controlling the volume envelope of an individual track or the entire mix.
- **Interest**. Making the mix special.

Many mixes have only four or five of these elements, but all six must be present for a great mix, as they are all equally important.

In some music genres that require simply re-creating an unaltered acoustic event (such as orchestral or jazz or any live concert recording), it’s possible that only the first four elements are needed to have a mix be considered great, but dynamics and interest have evolved to become extremely important elements even in those genres as modern tastes has evolved.

The Intangibles Of A Mix

It’s easy to think that getting a good mix is just a matter of pushing up some faders, getting a reasonable balance, adding some effects, and you’re finished. Although that might work for a rough mix, there are still a number of intangibles that are vitally important to a great mix. Awareness is always the first step in learning, so here are some things to consider before you start to move faders around.

The Arrangement

It's really easy to get caught up in just the audio portion of being an engineer, but unless you seriously consider how the music itself is put together (assuming that music is what you're engineering, of course), your ultimate product probably won't sound great no matter how good you are at balancing tracks.

Anyone with a little mixing experience has found that the arrangement is usually the #1 non-audio problem in a mix. In these days of unlimited tracks, it's all too easy to pile more and more musical elements along with double and triple tracks of everything you can think of. You can easily wind up with a hundred tracks to wade through before you know it, and that gives you an impossible task of making it sound like something more than a wad of dense audio goo. A good producer will usually bring some common sense to the arrangement, paring things down to where they're adding to and subtracting from the dynamics and intensity of the song, but sometimes the producer can also be the one demanding everything but the kitchen sink be added in. If that's the case, if the songwriter doesn't have an innate sense of arrangement (luckily many do), you've got a potential mixing mess on your hands.

That's why it's important that the mixing engineer be aware of some basic music-arrangement principles, which we'll cover in Chapter 5, "The Balance Element," because a big part of being a mixing engineer is knowing when to mute things and knowing just what elements need to take precedence at a certain part of the tune.

Good balance actually starts with good arrangement, so it's important to understand arrangement because so much of mixing is actually subtractive by nature. This means that the arrangement, and therefore the balance, is changed by the simple act of muting or lowering the level of an instrument whose part doesn't fit well with another and doesn't help the dynamic envelope of the song. If the instruments fit well together arrangement-wise so that they don't fight one another rhythmically or frequency-wise, then your life as a mixer just became immensely easier.

The Performances

Back in the ’50s, ’60s, and ’70s before recording and production techniques rose to the sophistication of today, a shaky performance by a player or singer could pretty much be accepted in the context of the song. With most players only getting a few takes at most to capture a part, the performance was evaluated more on feel than on precision. Today’s production techniques have evolved to the point where precision and feel are equally in demand during a performance, and as a result, the average listener has evolved to unconsciously expect that as well.

Shaky performances that are out of tune and rhythmically out of time can sometimes work if the mix is incredible, but usually only if the rhythm section is solid. One of the reasons why a mixing engineer has a huge box of tools, such as pitch correction, sound replacement, compression, and copy and paste editing, is to help fix or enhance performances so they’ll make the mix fit together better. That said, there’s a limit to how much a performance can be artificially helped before it sounds unnatural, which negatively impacts the mix as well. Like so many other elements given to the mixer, the better the performances, the better the mix will sound.

The Point Of Interest

Every song has something that's the main point of interest or something so compelling that you can't take your ears off it. If it doesn't, send the song back to the drawing board. It's not complete.

Although having control over the first five elements mentioned earlier may be sufficient for many types of audio jobs and might be just fine to create a decent mix, most popular music requires a mix that can take the song to another level. Although it's always easier with great tracks, solid arrangements, and spectacular playing, a great mix can take tracks that aren't particularly special and transform them into hit material so compelling that people can't hear enough of it. It's been done on some of your favorite all-time songs.

So how can we get to that point?

More than being just technically correct, a mix must be as interesting as a good movie. It must build to a climax while having points of tension and release to keep the listener subconsciously involved. Just as a film looks bigger than life, a great mix must sound bigger than real life. The passion and the emotion must be on a level where the listener is sucked in and forced to listen.

And the way to do that? Find whatever element is the most important to the song. In some cases (such as dance and hip-hop music), the most important element is the groove. Yet in other genres (such as country), it's the vocal, but in rock and pop it might be a signature line or hook or even the groove or the rhythm.

Even though the most important element is often the lead vocal, it doesn't necessarily have to be. It could be a riff, such as from the Rolling Stones' "Satisfaction" and "Start Me Up," or the intro to Coldplay's "Clocks," or the rhythm on the verses of Arctic Monkeys' "I Bet You Look Good on the Dance Floor," or Lady Gaga's "Born This Way." It's always a part so compelling that it forces you to listen to the song.

Which brings us to the nitty gritty of the book, where all the elements of a great mix are detailed even further in the next chapters.

Chapter 5

The Balance Element: The Mixing Part Of Mixing

The most basic element of a mix is balance. A great mix must start here first, because without balance, the other mix elements pale in importance. There’s more to balance than just moving some faders though, as we’ll see.

The Arrangement: Where It All Begins

Good balance starts with a good arrangement. It’s important to understand arrangement because so much of mixing is subtractive by nature. This means that the arrangement, and therefore the balance, is changed by the simple act of muting an instrument whose part either doesn’t fit well with another or doesn’t fit in a particular section of a song. If the instruments fit well together arrangement-wise so they help build the song dynamically and don’t fight one another frequency-wise, then the mixer’s life becomes immensely easier.

Tension And Release

All art is built around tension and release, which is just another expression for contrast. It's big against small, fat against slim, wide against narrow, and black against white. In photography it's shadows against light, in painting it's light against dark, in music it's loud against quiet, and in mixing it's full against sparse. That's what makes things interesting; you never know how big something is until you see something small to compare it to, and vice versa.

All good arrangements are filled with dynamic changes, which means loud versus quiet and full versus sparse. One of the jobs of a mixer is to create this tension and release when it's not there, and when it is, to emphasize it.

This is done by muting and unmuting tracks and changing the level of certain vocals or instruments at points within the song.

Conflicting Instruments

When two instruments occupy the same frequency band and play at the same volume at the same time, the result is a fight for attention. Think of it this way: You don't usually hear a lead vocal and a guitar solo at the same time, do you? That's because the human ear isn't able to decide which to listen to and becomes confused and fatigued as a result (see Figure 5.1).

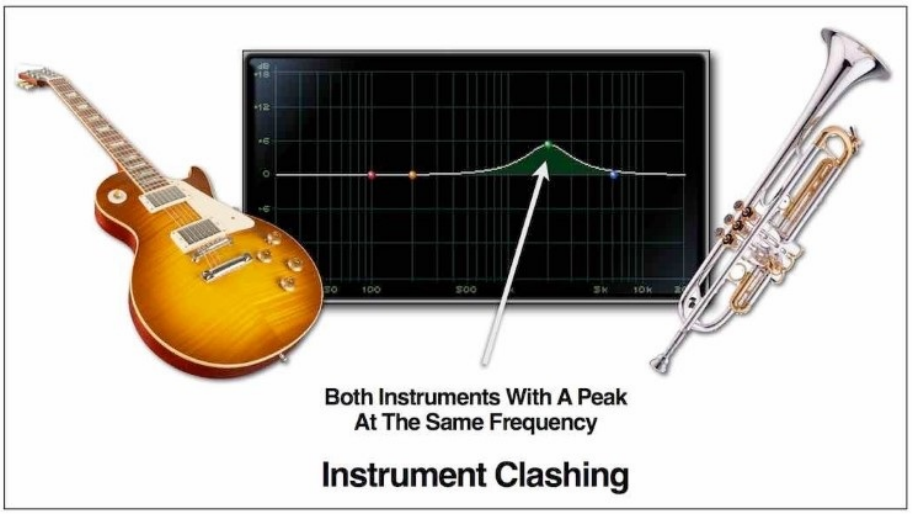

Figure 5.1: Conflicting instruments
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So how do you get around instruments conflicting with one another? First and foremost, a well-written arrangement keeps instruments out of each other's way right from the beginning. The best writers and arrangers have an innate feel for what will work arrangement-wise, and the result is an arrangement that automatically lays together without much help.

It's not uncommon to work with an artist or band that isn't sure of the arrangement or is into experimenting and just allows an instrument to play throughout the entire song, thereby creating numerous conflicts. This is where the mixer gets a chance to rearrange the track by keeping what works in the mix and muting any other conflicting instruments or vocals. The mixer can influence not only the arrangement this way, but also the dynamics and general development of the song as well.

To understand how arrangement influences balance, we have to understand the mechanics of a well-written arrangement first.

Most well-conceived arrangements are limited in the number of arrangement elements (not mix elements; there's a difference) that occur at the same time. An arrangement element can consist of a single instrument, such as a lead guitar or a vocal, or it can be a group of instruments, such as the bass and drums, a doubled guitar line, a group of backing vocals, and so on. Generally, a group of instruments playing exactly the same rhythm can be considered a single element. For

example: A doubled lead guitar or doubled vocal is a single element, as is a lead vocal with two additional harmonies. Two lead guitars playing different melody lines or chordal rhythms are two elements, however. A lead and a rhythm guitar are usually two separate elements as well.

Arrangement Elements

To understand how arrangements work, you have to understand what the basic arrangement elements are first. These may vary a little with genre, but the idea stays the same.

The Foundation

The foundation is usually comprised of the bass and drums, but can also include a rhythm guitar and/or keyboard if they're playing the same rhythmic figure as the rhythm section. Occasionally, as in the case of power trios, the foundation element will only consist of drums, since the bass usually needs to play a different rhythm figure to fill out the band's sound, so it therefore becomes its own element.

The Pad

A pad is a long sustaining note or chord that adds a sort of "glue" to the arrangement and therefore the mix. In the days before synthesizers, a Hammond organ provided the best pad, and was later joined by the Fender Rhodes. Synthesizers now provide the majority of pads, but real strings or a guitar power chord can also serve in that role as well.

The Rhythm

The rhythm element can come from any instrument that plays against the foundation element. That can mean a double-time shaker or tambourine, a rhythm guitar strumming on the backbeat, or congas playing a Latin feel. The rhythm element is used to add motion and excitement to the track. Take it away, and the track loses a bit of life and energy.

The Lead

A lead vocal, lead instrument, or solo.

The Fills

Fills generally occur in the spaces between lead lines or can be a signature line. You can think of a fill element as an answer to the lead. These elements can be seen visually in Figure 5.2.

ELEMENT	PURPOSE	TYPICAL INSTRUMENTS
Foundation	The instruments that provide the groove or pulse of the song	Drums and bass
Pad	Long sustaining notes that glue the mix elements together	Organ, electric piano, strings, guitar power chords
Rhythm	The instruments that provide motion to the song	Percussion, rhythm guitar
Lead	The focal point of the song	Lead vocal, lead or solo instrument
Fill	The instruments that fill in the spaces between the lead phrases	Solo instrument, background vocal

Figure 5.2: The arrangement elements

Arrangement Examples

To help you understand arrangements better, here are some examples of some hit songs and their arrangement elements.

"Born This Way" by Lady Gaga

To keep the song interesting, "Born This Way" utilizes song dynamics from section to section, adding and subtracting instruments and vocals to make the song more or less intense.

The five elements of the arrangement look like this:

- **The foundation.** As in most songs, it's the bass and drums.
- **The pad.** As in most dance songs, it's a synth pad that you can hear prominently in the first verse, but it's actually there for the entire song, adding the glue to the arrangement as well.
- **The rhythm.** This element utilizes an aggressive synth with a sawtooth wave shape that you can hear predominantly in the second half of the first verse.
- **The fills.** As in many hit pop songs, there's something in almost every space where there's not a vocal. In this case it's some sort of synth, but there are lots of sound effects as well.
- **The lead.** Gaga's vocals.

Listen to the following places for how the arrangement elements work together:

- ➡ The first half of the verse with just the vocal, pad, and rhythm.
- ➡ The first four bars of the second verse with just the vocal and kick drum. w The first half of the bridge with spoken word, synth bed, and sound effects.
- ➡ The first half of the first outro chorus with just vocals and kick drum.
- ➡ The last out-chorus vocal, where it breaks down to just the lead and harmony vocals.

"Rolling in the Deep" by Adele

The song begins with just the eighth-note guitar and lead vocal, which is joined by the kick drum in the second half of the verse. In the B-section, simple piano triads, the bass, and rest of the drums enter. See the development?

In the chorus, a strumming acoustic guitar and piano eighth notes push the song along, as does the entrance of the background vocal answers.

In the second verse, the low piano octaves on the “one and”; then the background vocals enter in the second B-section. See how the second verse develops? In the second chorus the background vocals add a harmony to make that section different from the first. The last verse and first out chorus break down to kick and fills.

The arrangement elements look like this:

- **The foundation.** Bass and drums.
- **The pad.** There’s not really a true pad, but in the B-section, the piano playing whole-note chords acts as a pad for a bit.
- **The rhythm.** The eighth-note guitar in the verse, the strummed acoustic and eighth-note piano in the chorus.
- **The lead.** The lead vocal.
- **The fills.** The background vocals and occasional clean lead guitar.

“Grenade” by Bruno Mars

The arrangement of the song follows a well-used formula where it develops from a sparse first verse to a big chorus, having a less sparse second verse, and finally peaking at the bridge. The tension is released by the stripped-down last outro verse, which is very unusual since most outros retain the intensity to the end.

The five elements of the arrangement look like this:

- **The foundation.** As is the norm, the foundation element is held down by the bass and drums.
- **The pad.** There’s an organ that plays just underneath everything that acts as the pad and glues the track together, which is, once again, pretty standard. You can never go wrong with an organ for this element.
- **The rhythm.** The arpeggiated electric piano line in the verse acts as the rhythm element, but during the chorus it switches to the double-time feel of the drums.
- **The lead.** As almost always, it’s the lead vocal.
- **The fills.** The fills are handled by the background vocals and the occasional percussion sound effect.

“Refugee” by Tom Petty and the Heartbreakers

As with most hits, the dynamics in “Refugee” are great; however, unlike many other songs, they’re not created by additional overdub layers, but by real dynamic playing by the band. The band’s playing breathes with the song, pushing it to a peak in the bridge and bringing it back down to a quiet third verse.

Let’s look at the arrangement elements:

- **The foundation.** As with most songs, the foundation element for “Refugee” is held down by the drums and bass.
- **The pad.** You can’t get a better pad element than a Hammond B-3, and that’s what you hear here.
- **The rhythm.** There’s a shaker that’s placed low in the mix so it’s not obvious, but it really pushes the song along with a double-time feel, and you’d miss it if it wasn’t there.
- **The lead.** Tom Petty’s lead vocal and Mike Campbell’s tasty guitar in the intro and solo.
- **The fills.** Once again it’s the guitar in the verses and the background vocal answers in the chorus.

The song builds and develops in a classic way. In the intro, the full band is playing with the lead guitar; then in the verse, it’s just the organ and rhythm section with rhythm guitar strums every four bars. In the last half of the verse, gets louder as the guitar kicks in, and in the chorus, the guitars go back to what they played in the intro (but they’re lower in the mix) and the background vocals answer the lead vocal.
The only thing fancy in this song is the doubled lead vocal in the bridge, and the fact that the first half of the solo is by the organ, followed by the guitar.

“What Hurts the Most” by Rascal Flatts

“What Hurts the Most” is a glowing example of the “new” country music in that it closely resembles layered pop music except for the addition of traditional country instruments such as fiddle, steel guitar, and banjo. As you would expect from a big-budget act, this song has absolutely state-of-the-art arranging, which is needed for a song with a simple form.

What’s especially cool is all the sections of the song that repeat but are slightly different the second or third time through. A good example is the line in the last bar of the first half of the intro, which is first played on acoustic guitar, then doubled with the fiddle the second time through. On the third pass there’s a steel guitar fill.

The second verse develops with the entrance of fiddle and electric guitar; then in the second chorus, the steel guitar and banjo enter. The last chorus is where the song stops and the melody changes as the background vocals enter right afterwards, which keeps your interest because you've not heard it this way in the song before.

Let's look at the elements:

- **The foundation.** Bass and drums
- **The pad.** Steel and big electric guitar chords during the chorus.
- **The rhythm.** Acoustic guitar in the verses and the banjo and shaker in the choruses.
- **The lead.** Fiddle in the intro and interlude, lead vocal in the verses and choruses, and lead guitar in the solo.
- **Fills.** Steel guitar answers in the interlude and background vocal answers in the last chorus.

Rules For Arrangements

There are a couple of easy-to-remember rules that will always make even the densest arrangement manageable.

1. **Limit the number of elements.** Usually there should not be more than four arrangement elements playing at the same time. Sometimes three elements can work very well, but very rarely will all five elements simultaneously work.

***TIP**: More than five elements occurring simultaneously confuses the listener and causes listening fatigue.*

2. Everything in its own frequency range. The arrangement, and therefore the mix, will fit together better if all instruments sit in their own frequency range. For instance, if a synthesizer and rhythm guitar play in the same octave, they may clash frequency-wise and fight each other for attention. The solution would be to change the sound of one of the instruments so they fill different frequency ranges, have one play in a different octave, or have them play at different times but not together.

   "So much of mixing is what you take away, either level-wise or frequency-wise. There are so many things that you have to eliminate in order to make it all sit and work together."

   —Lee DeCarlo

***TIP**: Here are some ways to prevent instrument fighting:*

- *Mute the offending instruments so that they never both play at the same time.*
- *Lower the level of the offending instrument.*
- *Tailor the EQ so that the offending instrument takes up a different frequency space.*
- *Pan the offending instrument to a different location.*
- *Change the arrangement and re-record the track.*

Where To Build The Mix From

Different mixers start from different places when building their mix. This has as much to do with their training and experience as with the type of material being mixed. For instance, many old-time New York mixers and their protégés start from the bass guitar and build the mix around it. Some work on the vocal first, since it's the most important element of a song. Some mixers work from the drum overheads first, tucking in the other drums as they go along, while others work from the toms. Many mixers mix with everything up, only soloing specific instruments that seem to exhibit a problem. Still others are completely arbitrary, changing the starting place from song to song depending upon whatever instrument needs to be focused on. And of course, so many mixers these days begin with the kick drum.

> *"I start with everything on, and I work on it like that. The reason is that, in my opinion, the vocal is going to be there sooner or later anyway. All the instruments are going to be there sooner or later, so you might as well just get used to it. And I think that's also what helps me see what I need to do within the first passage."*
>
> *—Jon Gass*

> *"It really is like building a house. You've got to get the foundation of bass and drums and then whatever the most important part of the song is, like the vocalist, and you've got to build around that. I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom. I build the drums on top of that. After I do the bass and drums, then I get the vocal up and then build everything from there."*
>
> *—Benny Faccone*

> *"I start with the drums and bass and get the basic foundation, then add the guitars and piano or anything that would be considered part of the rhythm section. After that feels good, then I put a vocal in, because the style of music that I do is all vocal-driven, so the sooner I get it in the mix, the better. After that, I place any of the ear candy around the vocal and rhythm section."*
>
> *—Bob Bullock*

> *"I'd love to say that I always build it from the vocal, but usually what I'll do is deal with the drums to get them to act like one fader's worth of stuff instead of 20 or whatever it is. Once I've done that, everything*

seems to come up at once."

—Andrew Scheps

"It's totally dependent on the music, but if there was a method of my approach, I would say the rhythm section. You usually try to find the motor and then build the car around it."

—Bruce Swedien

"I typically bring in the overheads first because my overheads are overall drum mics, then I fit the snare into that, then I get the kick happening. At that point I take a look at the toms and see if they're really adding a lot of good stuff to the drum sound. I'll just keep them on and set them where I want and then push them for fills, then tuck in the hat."

—Allen Sides

Wherever you start from, it's generally agreed that the vocal, or whatever is the most prominent or significant melody instrument, has to make its entrance into the mix as soon as possible. The reason for this is twofold. First of all, the vocal is probably going to be the most important element, so it will need to take up some of the frequency space that already may be assigned to other supporting instruments. If you wait until late in the mix to put the vocal in, there may not be enough space left, and the vocal will never sit right with the rest of the track.

The second reason has to do with effects. If you tailor all of your effects to the rhythm section and supporting instruments, there may be none left when it's time to add in the vocal or most prominent instrument.

What's The Genre?

The type of program being mixed will frequently have an effect on where you build the mix from. For instance, when mixing dance music where the kick is everything, that's the obvious choice for a starting point. Mixing an orchestra, however, requires a different technique. According to Don Hahn, "The approach is totally different because there's no rhythm section, so you shoot for a nice roomy orchestral sound and make it as big a sound as you can get with the amount of musicians you have. You start with violins, then violas if you have them, cellos, then basses. You get all that happening and then add woodwinds, French horns, trombones, trumpets, and then percussion."

In jazz, the melody instrument can be the starting point with the bass inserted shortly after to solidify the foundation. Early pop and rock engineers started with the bass and built around that, although that's pretty much changed to the kick drum these days.

Typical Mix Starting Places

From the bass
From the kick drum
From the snare drum
From the toms
From the overheads
From the lead vocal or main instrument
When mixing a string section, from the highest string (violin) to the lowest (bass)

Level Setting Methods

Setting levels by using the meters has been debated from the beginning of mixing time. Some mixers feel that they can get in the ballpark by setting the levels with the master mix meters alone, while others discount any such method out of hand. The fact of the matter is that for those using the meter method, feel and instinct are still a large part of their technique, making it equally as valid as those who rely on instinct alone.

Regardless of the level you begin at, keep watching your master buss meters for overloads with the entrance of each instrument to be sure that a clean signal path is maintained.

As with everything else that you read, try the following methods, use what works, and throw away the rest.

> *"I usually start with the bass at about -5 (on a VU meter) and the kick at about -5. The combination of the two, if it's right, should hit about -3 or so. By the time the whole song gets put together and I've used the computer to adjust levels, I've trimmed everything back somewhat. The bass could be hitting -7 if I solo it after it's all done."*
>
> *—Benny Faccone*

> *"I'll start out with the kick and bass in that area (-7 VU). By the time you put everything else in, it's +3 anyway. At least if you start that low you have room to go."*
>
> *—Don Smith*

> *"Usually a good place to start is the kick drum at -6 or -7 or so. I'll try to get a bass level that is comparable to that. If it's not exactly comparable on the meter because one's peaking and one's sustaining, I get them to at least sound comparable. That's kind of a good starting place for me."*
>
> *—Ed Seay*

> *"I'll get the snare drum constantly hitting the back beat of the tune at around 5, and everything gets built around it."*
>
> *—Lee DeCarlo*

***TIP**: One of the tried-and-true methods is to begin with the first instrument hitting about*

-10dBFS on a typical digital peak meter. That provides enough headroom so that when additional instruments enter, the master buss meter won't immediately be lighting the overload indicator.

Reference Mixes

As said back in Chapter 3, one way to make sure that your mix stays on track is to constantly refer to one or more reference songs during your mix. This allows you to not only check for element volume balances, but also frequency balance against a reference that you know sounds great.

An easy way to do that is with one of comparison plugins like SampleMagic's Magic AB (see Figure 5.3), which not only allows you to instantly compare your mix with up to 9 other tracks, but perfect match the level between them all and your mix, as well as pick a specific section of the song to compare it too. Similar plugins include Nugen Audio A|B Assist and Melda Production MCompare.

Figure 5.3: SampleMagic Magic AB

Chapter 6

The Panorama Element: *Placing the Audio In The Soundfield*

One of the most overlooked or taken for granted elements in mixing is panorama, or the act of placing a sound element in the soundfield. To understand panorama we must first understand that the stereo sound system (which is two separate audio channels, each with its own speaker) represents sound spatially. Panning lets us select where in that space we place the sound.

In fact, panning does more than just that. Panning can create excitement by adding movement to the track and adding clarity to an instrument by moving it out of the way of other sounds that may be clashing with it. Correct panning of a track can also make a recording sound bigger, wider, or deeper, sometimes all at the same time.

The Stereo Picture

What is the proper way to pan? Are there any rules? Just like so many other things in mixing, there's a method to follow that has some good reasoning behind it, although it may sometimes seem that panning decisions can be pretty arbitrary.

Imagine that you're at the movies and watching a Western. This scene is a panorama of the Arizona desert, and right in the middle of the screen is a cowboy sitting on his horse in a medium shot from his boots up. Now a pack of Indians (we'll say six) are attacking him, but we can't see them because the cowboy is in the shot directly in front of them (see Figure 6.1, but with singers instead). If we can't see them, their impact as a suspense builder is really limited, not to mention the fact that they cost the production money that just went to waste. Wouldn't it be better if the director moved the Indians to the left, out from behind the cowboy, so we could see them? Or maybe even spread them out across the screen so the attack seems larger and more intimidating?

Of course, that's what we do with the pan pot. It gives the mixer (the director) the ability to move the background vocals (the Indians) out of the way of the lead vocal (the cowboy) so that in this case we can hear (see) each much more distinctly.

Figure 6.1: Panning visualization
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The Phantom Center

Stereo, which was invented in 1931 by Alan Blumlein at EMI Records (the patent wasn't renewed in 1959 when the format was taking off—d'oh!), features a phenomenon known as the phantom center. The phantom center means that the output of the two speakers combine to provide an imaginary speaker in between (see Figure 6.2). This imaginary image can sometimes shift as the balance of the music shifts from side to side, which can be very disconcerting to the listener, especially if the speakers are placed far apart. As a result, film sound has always relied upon a third speaker channel in the center to keep the sound anchored. (This format is called LCR, for Left, Center, Right). This third channel never caught on in music circles however, mostly because consumers had a hard enough time finding a place for two speakers, let alone three.

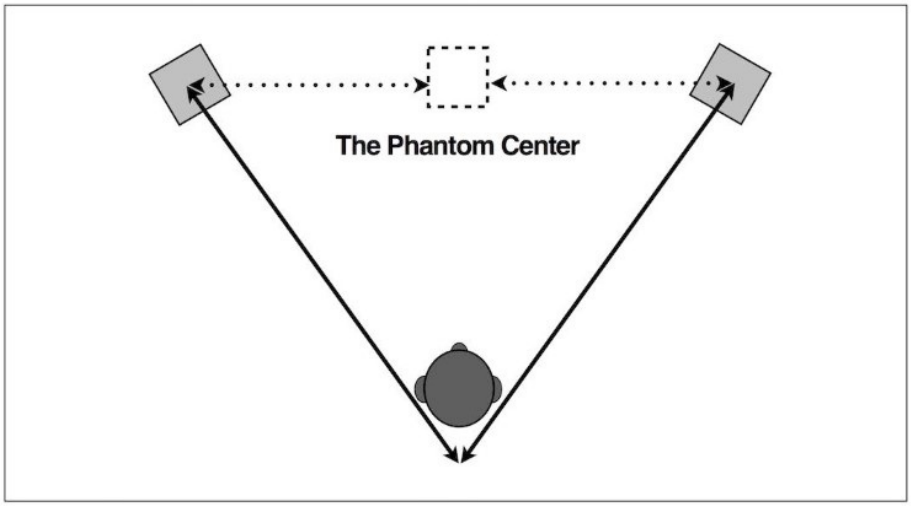

Figure 6.2: The phantom center
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The Big Three

That said, there are three major panoramic areas in a mix: the extreme hard left, the extreme hard right, and the center (see Figure 6.3).

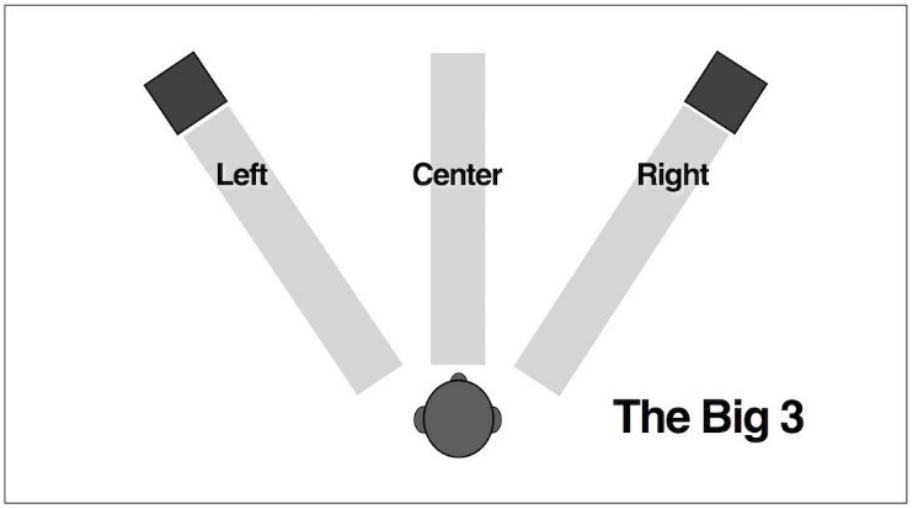

Figure 6.3: The big three panning areas
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The center is obvious in that the most prominent music element (like the lead vocal) is usually panned there, but so is the kick drum, bass guitar, and even the snare drum. Although putting the bass and kick up the middle makes for a musically coherent and generally accepted technique, its origins come really from the era of vinyl records.

With the first stereo recordings in the mid-'60s, sometimes the music from the band was panned to one channel while the vocals were panned to the opposite one. This was because stereo was so new that the recording and mixing techniques for the format hadn't been refined yet, so what we now know as pan pots weren't available on mixing consoles. Instead, a three-way switch was used to assign the track to the left output, the right output, or both (the center). Sometimes the channels were even dedicated to either left or right, like on the famous REDD console at Abbey Road Studios that was used to record The Beatles and other major acts of the day.

Because music elements tended to be hard panned to one side, this caused some major disc-cutting problems in that if any low-frequency boost was added to the music on just that one side, the imbalance in low-frequency energy would cause the cutting stylus to cut right through the groove wall when the master lacquer disc (the master record) was cut. The only way around this was to either decrease the amount of low-frequency energy from the music to balance the sides or pan the bass and kick and any other instrument with a lot of low-frequency information to the center. In fact, a special equalizer called an Elliptical EQ was used during disc cutting specifically to move all the low-frequency energy from both sides to the center to prevent any cutting problems.

Likewise, as a result of the vast array of stereo and pseudo-stereo sources and effects that came on the market over the years, mixers began to pan these sources hard left and right almost without even thinking, since a mixer's main task is to make things sound bigger and wider. Suddenly things sounded huge! The problem came later, when almost all keyboards and effects devices came with stereo outputs (many are actually pseudo-stereo with one side just chorused a little sharp and then flat against the dry signal). Now the temptation was to pan all of these "stereo" sources hard left and right on top of one another. The result was "Big Mono."

> *"I think that there are three sacred territories in a mix that if you put something there, you've got to have an incredibly good reason. That's extreme left, the center, and extreme right. I've noticed that some mixers will get stereo tracks from synthesizers and effects, and they just instinctively pan them hard left and hard right. What they end up with is these big train wrecks out on the ends of the stereo spectrum. Then they pan their kick, snare, bass, and vocals center, and you've got all this stuff stacked on top of each other. If it were a visual, you wouldn't be able to see the things behind the things in front."*
>
> *—Dave Pensado*

Big Mono

Big Mono occurs when you have a track with a lot of pseudo-stereo sources that are all panned hard right and hard left. In this case, you're not creating much of a panorama because everything is placed hard left and right, and you're robbing the track of definition and depth because all of these tracks are panned on top of one another (see Figure 6.4).

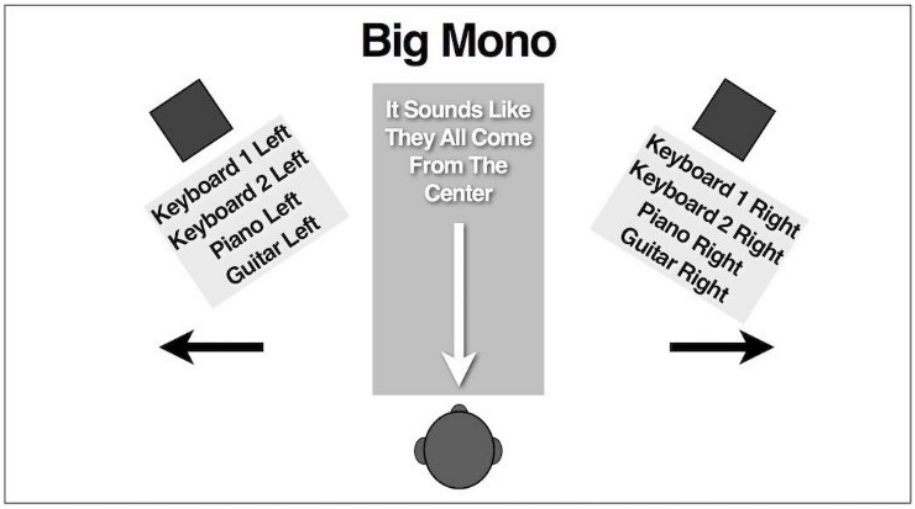

Figure 6.4: Big Mono
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> *"One of the things I don't like is what I call 'Big Mono,' where there's no difference in the left and the right other than a little warble. If you pan the left and right wide, and then here comes another keyboard and you pan that left and right wide, and then there's the two guitars and you pan them left and right wide, by the time you get all this stuff left and right wide, there's really no stereo in the sound. It's like having a big mono record, and it's just not really aurally gratifying. So to me, it's better to have some segregation, and that's one of the ways I try to make everything heard in the mixes. Give everybody a place on the stage."*
>
> *—Ed Seay*

The solution here is to throw away one of the stereo tracks (the chorused one; keep the dry channel) and make your own custom stereo patch with either a pitch shifter or a delay (See Chapter 8, "The Dimension Element," for how this is done.) Instead of panning the two channels hard left and right, find a place somewhere inside those extremes. One possibility is to pan the left source to about 10:00 while the right is panned to about 4:00. Another more localized possibility would be to put the left to 9:00 and the right all the way to 10:30. This gives the feeling of localization without getting too wide.

***TIP**: Panning narrowly on the same side (like 1:30 and 3:00) provides the width and*

depth of stereo while still having precise localization within the stereo soundfield.

Panning Outside The Speakers

Some mixers like to use the phantom images afforded by effects processing to pan an instrument outside the speakers. In this case, the phase differences make the instrument seem to come from outside the speakers instead of from them. While some find this effect disconcerting, it can be very effective under the right circumstances.

There are two ways to accomplish this:

1. On a stereo instrument, flip the phase of one channel. The panning will now seem to be beyond the speakers.

2. On a stereo instrument, feed some of the right-channel signal to the left channel out of phase, and feed some of the left-channel signal to the right channel out of phase. This can be done easily by copying both channels, flipping the phase on both, panning them opposite of the way they're already panned, and gradually increasing the level (see Figure 6.5). As the level increases, the sound will seem to pan outside the speakers.

Or, use one of the many plugins that offers a Mid/Side or M/S feature (check out the Brainworx products, which all have this feature), which can provide even more control than the manual method.

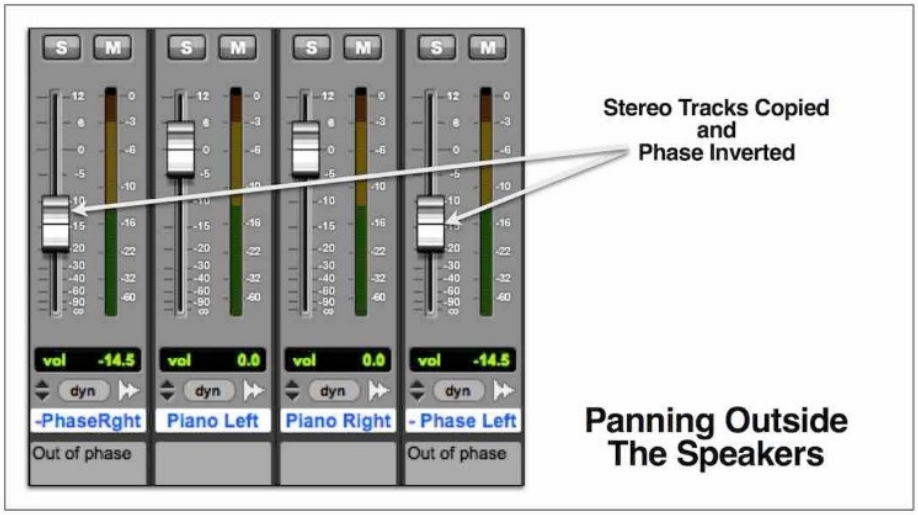

Figure 6.5: Setup for panning outside the speakers

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Beyond Panning For Placement

While panorama usually relates only to placement of a sound source within the stereo soundfield, there are actually other areas where panning can help a mixer.

Some mixers find that after their panning placements have been chosen and all their EQing is complete, sometimes just moving the instrument's pan slightly can actually provide more clarity to the mix. In fact, if instruments are fighting for frequency space, the first thing to try is alternate panning.

Some mixers even do some of their panning in mono (yes, that's right!). This is because it's easy to hear differences in phase, which make the instrument stand out when you go back to stereo.

> *"I check my panning in mono with one speaker, believe it or not. When you pan around in mono, all of a sudden you'll find that it's coming through now and you've found the space for it. If I want to find a place for the hi-hat, for instance, sometimes I'll go to mono and pan it around, and you'll find that it's really present all of a sudden, and that's the spot."*
>
> *—Don Smith*

***TIP**: When mixing a dance club record, it's best not to pan important elements very wide, since the speakers can be on different sides of the dance floor so half the people won't hear it. It's best to keep important elements either up the middle or maybe at 10:30 and 1:30.*

Panning In Surround Sound

Surround-sound formats offer increased enjoyment to the listener but increased complexity to the mixer. There are numerous speaker formats, all requiring a different panning approach from stereo. Let’s look at them.

A Bit Of History

Surround sound has actually been with us in one form or another for more than 50 years. Film has always used the three-channel “curtain of sound” developed by Bell Labs in the early ‘30s since it was discovered that a dedicated center channel provided the significant benefit of anchoring the center by eliminating “phantom” images (in stereo, the center images shift as you move around the room) and provided better frequency response matching across the soundfield.

The addition of a rear effects channel to the front three channels dates as far back as 1941 with the “Fantasound” four-channel system utilized by Disney for the film Fantasia, and then again in the 1950s with Fox’s Cinemascope, but it didn’t come into widespread use until the ’60s, when Dolby Stereo became the de facto surround standard. This popular film format uses four channels (left, center, right, and a mono surround, sometimes called LCRS) and is encoded onto two tracks. Almost all major shows and films currently produced for theatrical release and broadcast television are available in Dolby Stereo because it has the added advantage of playing back properly in stereo or mono if no decoder is present, which makes it compatible with a wide variety of theater sound systems.

With the advent of digital delivery formats capable of supplying more channels in the ‘80s, the number of rear surround channels was increased to two, and a low-frequency effects channel was added to make up the six-channel 5.1, which soon became the modern standard for most films (the Sony SDDS 7.1 eight-channel system being the exception), music, and digital television. Today we’ve graduated to far more advanced formats, such as 7.1, 11.2, and the totally revolutionary multi-speaker Dolby Atmos system.

And of course, there’s always the four-channel Quad from the 1970s, the music industry’s attempt at multichannel music that killed itself as a result of two non-compatible competing systems (a preview of the Beta vs. VHS videotape war that was soon to come), both of which suffered from an extremely small sweet spot.

The LFE Channel

LFE stands for low-frequency effects and is sometimes referred to in film production circles as the “Boom” channel because that’s what it’s there for—to enhance the low frequencies of a film so you get the extra boom effect from an earthquake, plane crash, explosion, or other such dramatic scene requiring lots of low frequencies.

The LFE channel, which has a frequency response from about 30Hz to 120Hz, is unique in that it has an additional 10dB of headroom built into it. This is needed to accommodate the extra power required to reproduce the low- frequency content without distortion.

Bass Management

The bass manager (sometimes called bass redirection) is a circuit that takes all the frequencies below 80Hz from the main channels and the signal from the LFE channel and mixes them together into the subwoofer (see Figure 6.6). The reason why this is done is to make use of the subwoofer for more than the occasional low-frequency effect, since it’s being used in the playback system already. This enables the effective response of the entire playback system to be lowered to about 30Hz if the subwoofer is large enough.

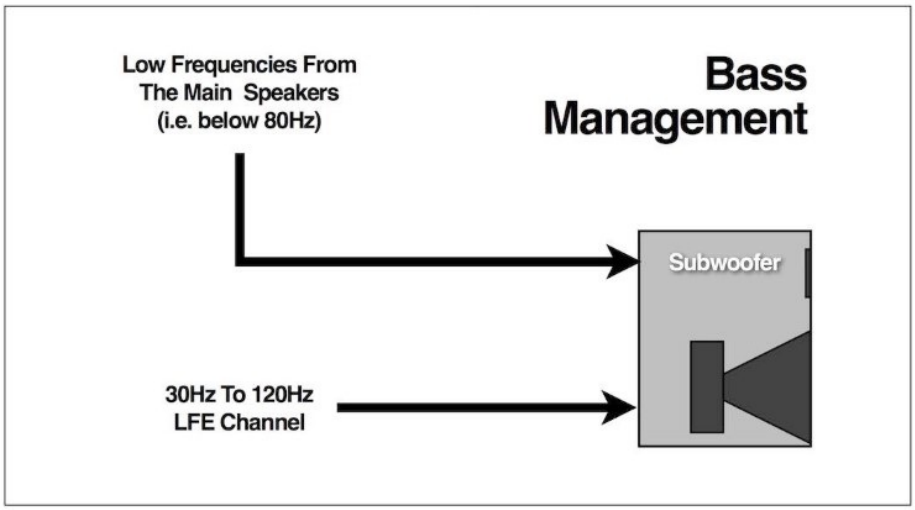

Figure 6.6: The LFE channel
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Since the overwhelming majority of consumer surround systems (especially the average inexpensive ones) contain a bass-management circuit, if you don’t mix with one then you may not be hearing things the way the people at home are. And, since the bass manager provides a low-frequency extension below that of the vast majority of studio monitors, the people at home may actually be hearing things (such as unwanted rumble) that you can’t hear while mixing.

Surround Sound Formats

Before we can discuss the panning technique for surround sound, it's important to be familiar with the various formats that it's available in. In the various configurations, the first number designates the number of main channels. If a low-frequency effects channel is used, the channel designation becomes "x.1"; if a low-frequency effects channel isn't included in the format, it becomes "x.0".

Three Channel (3.0)

This is an early Dolby Surround encoding format utilizing stereo front speakers and a mono speaker in the rear for surround (see Figure 6.7). This system was first introduced in 1982 as a way to encode surround information onto CDs and VHS videotapes.

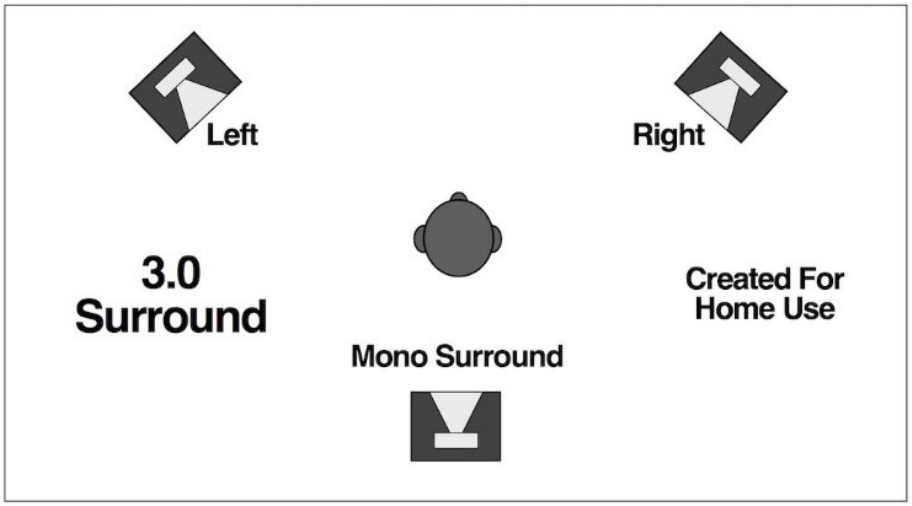

Figure 6.7: 3.0 surround
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LCRS Four Channel (4.0)

LCRS stands for Left, Center, Right, Surround and is the basic cinema setup of three speakers behind the screen fed by separate channels, and a single surround channel that feeds multiple speakers throughout the theater (see Figure 6.8).

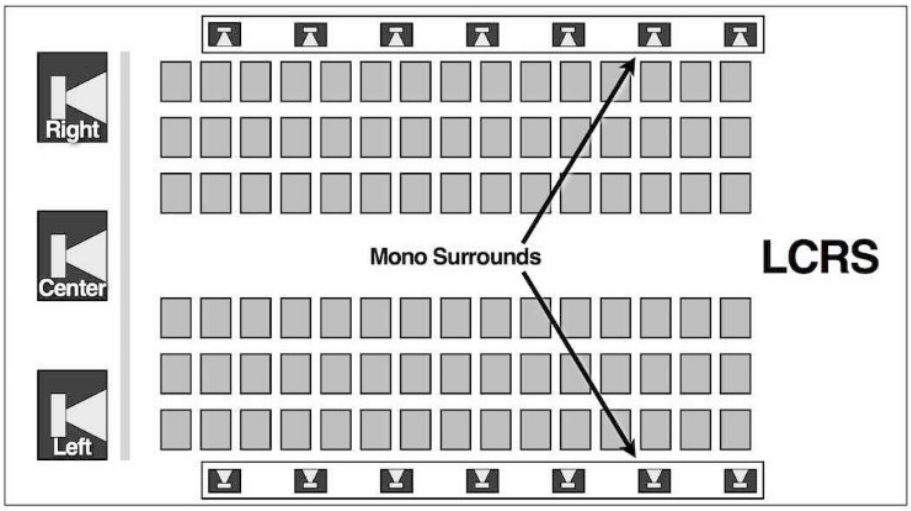

Figure 6.8: LCRS four channel surround
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Quadraphonic Four Channel (4.0)

The first true consumer surround format called “Quadraphonic” (or just “Quad” for short) was introduced in the early ’70s. It basically consisted of a stereo pair in front of the listener and another stereo pair behind him. The format never caught on primarily due to the fact that there were two competing delivery methods, which caused consumers to hesitate purchasing for fear they’d pick the one that didn’t win (see Figure 6.9).

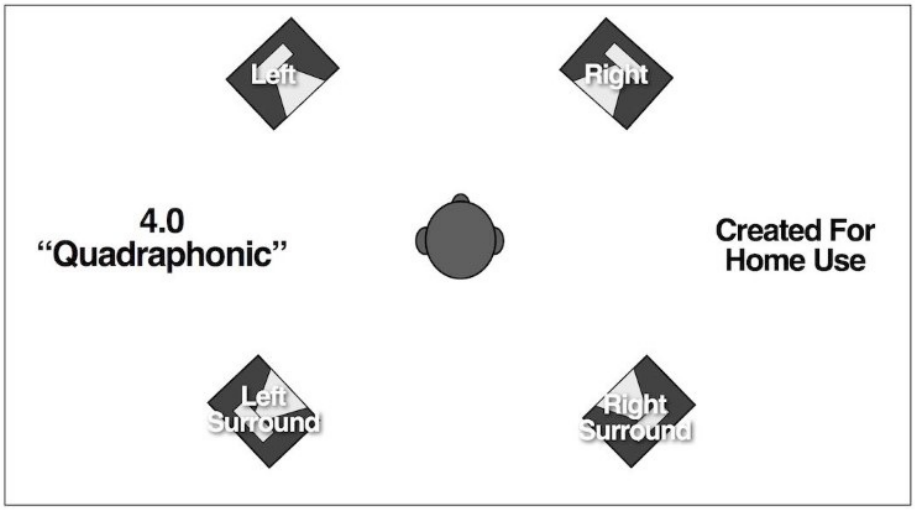

Figure 6.9: Four channel quad surround
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Five Channel (5.0)

Dolby developed its Pro Logic encoding especially for delivery of multichannel audio to the home, and gradually the number of channels it could encode evolved from three-channel (stereo with a single surround), to four-channel (LCRS with a single surround), and finally to five-channel (LCRS with stereo surrounds). This became a popular format for television-show audio, especially when it became possible to deliver it digitally (see Figure 6.10).

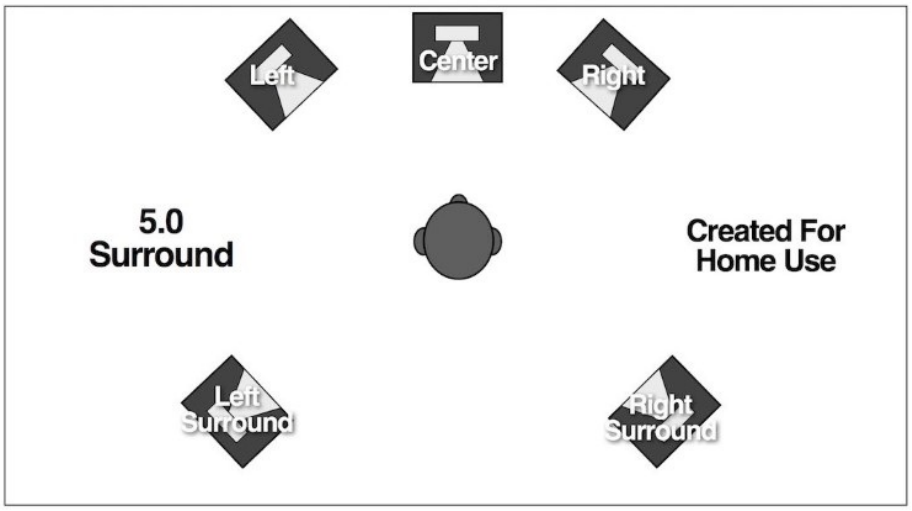

Figure 6.10: 5.0 surround
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5.1

The six-channel 5.1 surround was the gold standard for many years in both theater sound and home theater. The format consists of three channels across the front (L,C,R), two surround channels (left surround or Ls, and right surround or Rs), and a low-frequency effects channel (LFE) (see Figure 6.11).

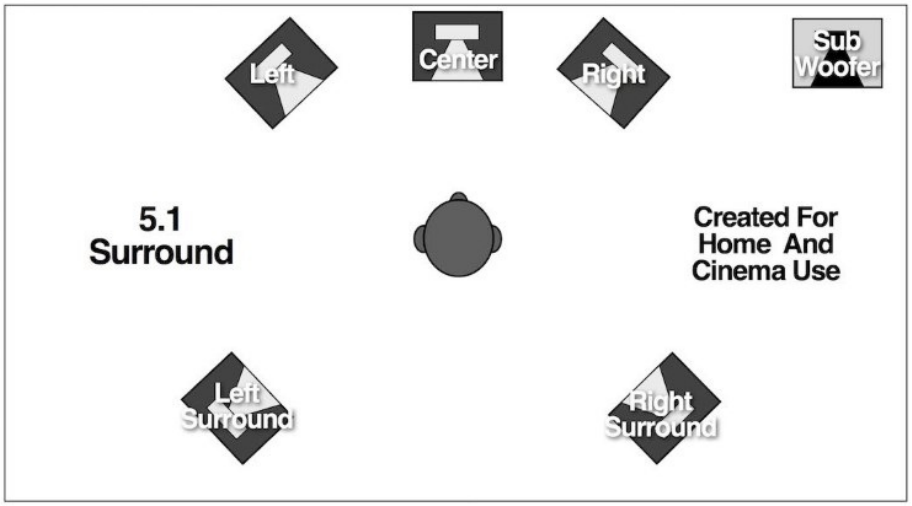

Figure 6.11: 5.1 surround
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6.1

Even though 5.1 was the standard for theater surround for some time, many film mixers complained that the 5.1 format was too limiting and they couldn't easily localize effects in the house with only two surround channels. The 6.1 format offers a center surround channel in an effort to improve the localization (see Figure 6.12).

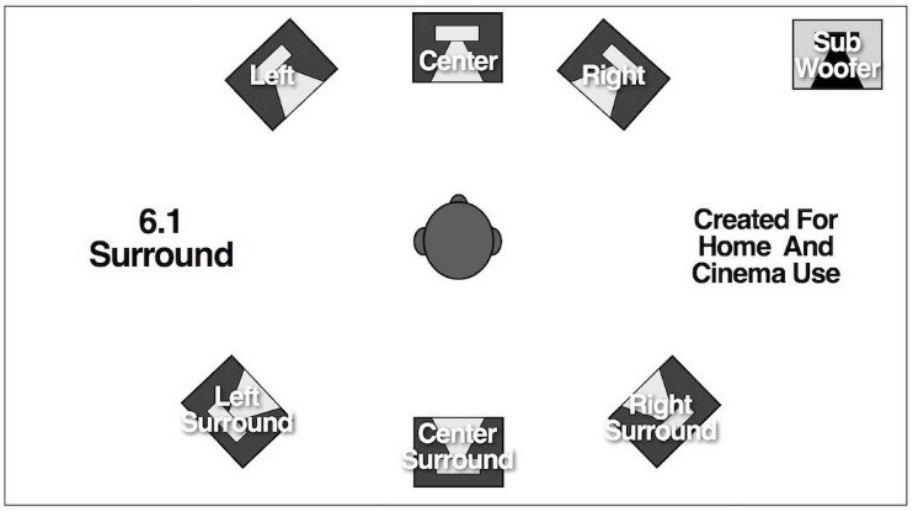

Figure 6.12: 6.1 surround
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7.1 SDDS

For a brief time, Sony offered their own eight-channel format known as Sony Dynamic Digital Sound, or SDDS. This was a 7.1 format that was configured somewhat contrary to what you might expect. Instead of additional surround channels, five channels were used across the front. One of the reasons for the format is that it provided the higher sound pressure level needed for larger theaters, but many also argued that it provided increased panning precision across the front speakers as well (see Figure 6.13).

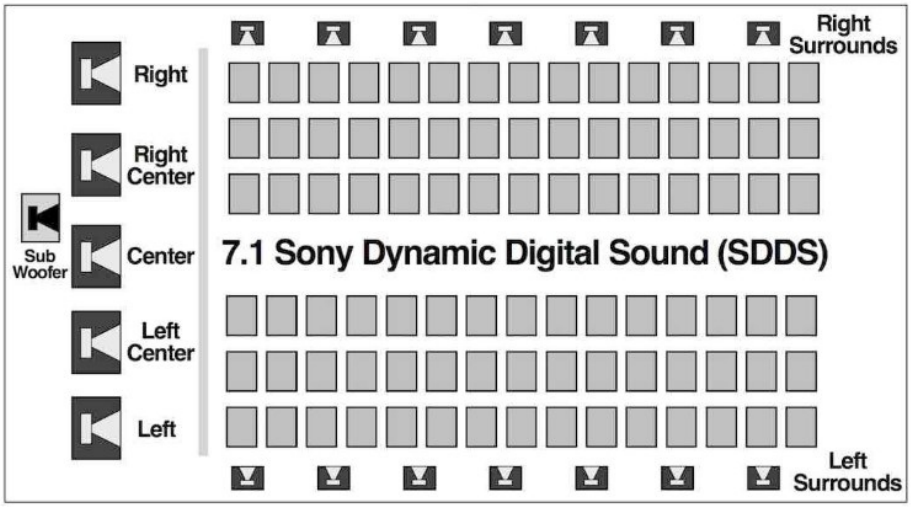

Figure 6.13: 7.1 SDDS surround
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7.1

As both listeners' and mixers' tastes became more sophisticated, it was determined that the stereo rear channels of 5.1 weren't enough of a sonic immersion to satisfy moviegoers or home-theater lovers. The eight-channel 7.1 attempts to change that with the inclusion of additional surround channels on the sides (see Figure 6.14).

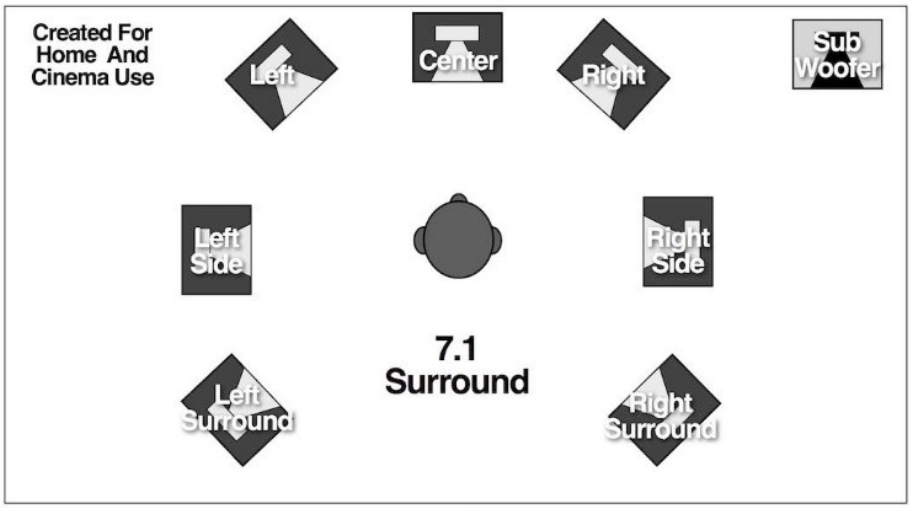

Figure 6.14: 7.1 surround
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10.2

Tomlinson Holman originally coined the term "5.1" when he was the chief scientist at Lucasfilm, and his TMH Labs took surround much further than the rest of the industry. While most firms were still experimenting with 5.1, TMH was far ahead when they created the 10.2 format, which is the same as the 7.1 format except for the addition of stereo LFE channels, a center rear channel, and stereo height channels placed in the front over the screen (see Figure 6.15). As an aside, Tom is also responsible for the THX sound standard that you see in theaters; it stands for "**T**om **H**olman e**X**periment."

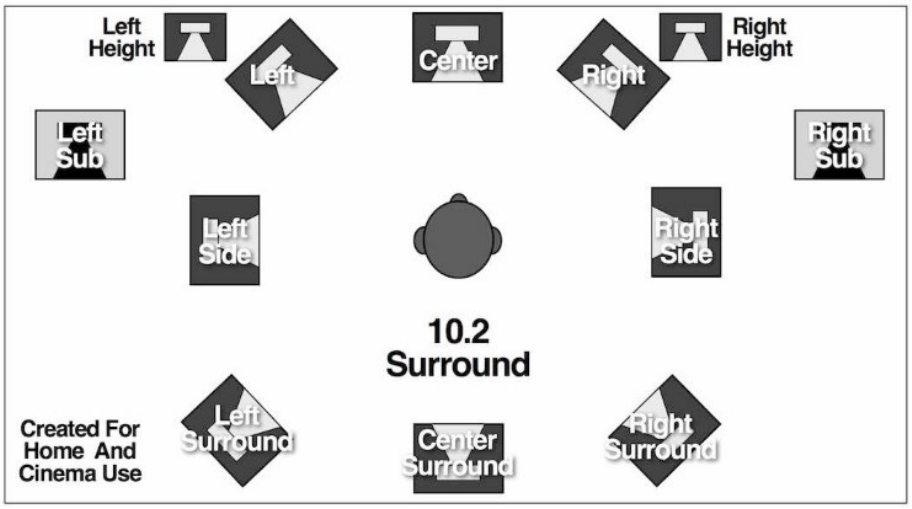

Figure 6.15: 10.2 surround
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11.1

In an attempt to totally immerse the listener in sound, the 12-channel 11.1 system was developed. This format adds additional side surround channels as well as stereo height channels above the screen (see Figure 6.16).

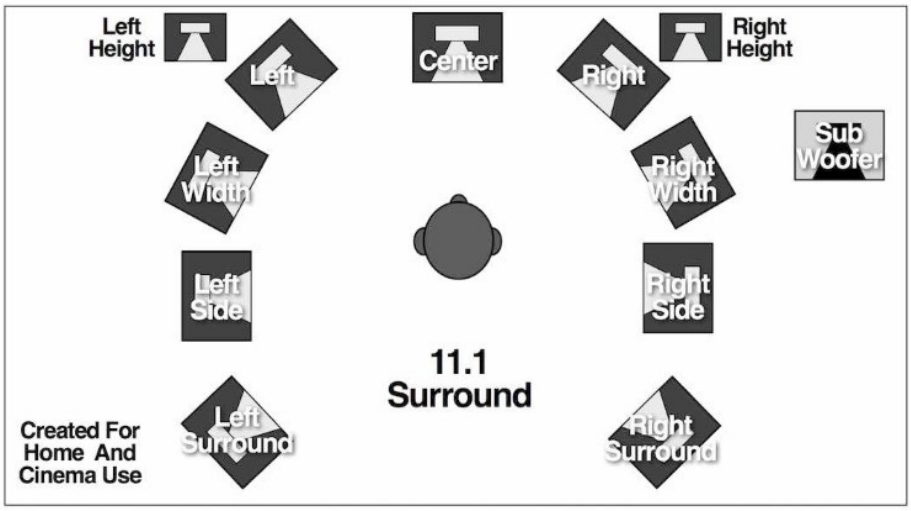

Figure 6.16: 11.1 surround
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Dolby Atmos

Dolby's Atmos playback system takes listener immersion and mixer panning ability to a whole new level. The system features up to 64 channels, utilizing additional screen channels, surround channels, an array of ceiling channels, and surround subwoofers. This gives a mixer an unprecedented ability to pan virtually anywhere around the listener. What's more, a single Atmos file contains all the metadata to automatically downmix it any other format, even stereo (see Figure 6.17).

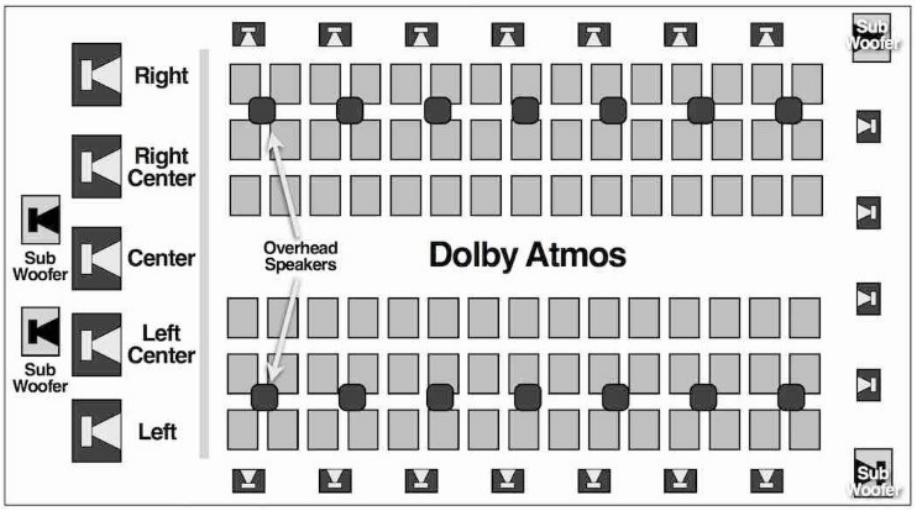

Figure 6.17: Dolby Atmos speaker placement
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Differences Between Surround For Picture And For Music

Normally in the theater, all of the primary sound information comes from the front speakers, and the surround speakers are utilized mostly for ambience info in order to keep your attention on the screen. The LFE is intended to be used just for special effects such as explosions and earthquakes and is therefore used infrequently. One of the reasons that the surround speakers don't contain more source information is a phenomenon known as the "exit-sign effect," which means that your attention is drawn away from the screen to the exit sign over the emergency exit at the side of the screen when the information from the surrounds is too loud or the front panning is too wide.

Music-only surround sound has no screen to focus on and therefore no exit-sign effect to worry about. Take away the screen, and it's now possible to utilize the surround speakers for more creative purposes.

Surround Mixing Schools Of Thought

There are two schools of thought about how surround sound for music should be mixed. The “audience” method puts the music in the front speakers and the hall ambience in the surrounds, just as if you were sitting in the audience of a club or concert hall. This method may not utilize the LFE channel at all and is meant to reproduce an audience perspective of the musical experience from what would arbitrarily be called “the best seat in the house.”

The second method is the “middle of the band” method. In this case the musical instruments are spread all over the room via the main channels, and that puts the listener in the center of the band and envelopes him with sound. This method usually results in a much more dramatic soundstage that is far bigger-sounding than the stereo that we’re used to. Middle-of-the-band placement may not be as authentic a soundscape as some music (such as any kind of live music where the listener’s perspective is from the audience) might require, however.

What Do I Put In The Center Channel?

In film mixing, the center channel is used primarily for dialog so the listener doesn't get distracted by movement in the soundfield. In music, however, its use prompts debate among mixers.

No Center Channel

Many veteran engineers who have mixed in stereo all their lives have trouble breaking the stereo paradigm to make use of the center channel. These mixers continue to use a phantom center from the left and right front speakers, or prefer to use the center speaker as a height channel or not use it at all (see Figure 6.18).

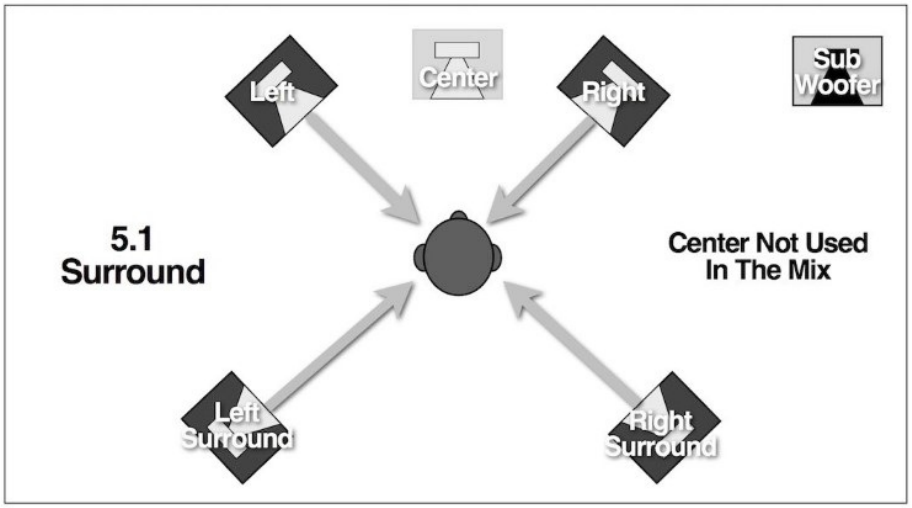

Figure 6.18: No center channel
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Isolated Elements In The Center Channel

Many mixers prefer to use the center channel to isolate certain elements, such as lead vocals, solos, and instruments with high-frequency information that infrequently appears. While this might work in some cases, many times the isolated elements seem disconnected from the rest of the soundscape unless they're bled into the other channels (see Figure 6.19).

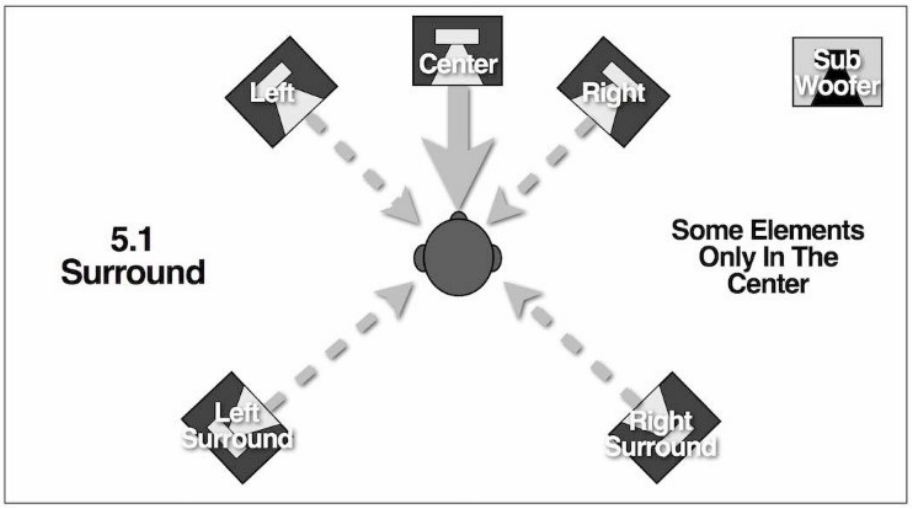

Figure 6.19: Isolated center channel
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The Center As Part Of The Whole

Mixers who use the center channel to its fullest find that it can act to anchor the sound and eliminate any drifting phantom images. In this case, all front and rear speakers have equal importance, with the balance changing the sound elements placed in the soundscape (see Figure 6.20).

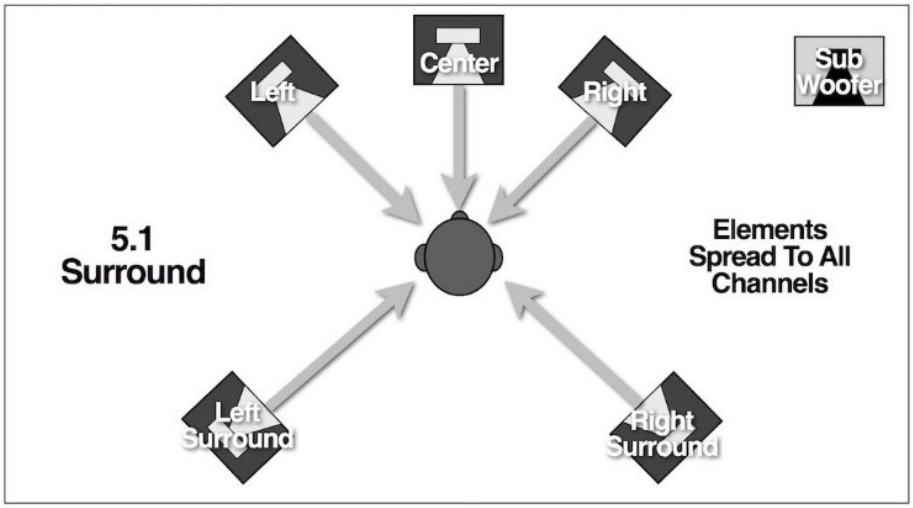

Figure 6.20: Center as part of the whole
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> *"I guess if I were to encapsulate the rule, the things that I used to put in the middle I put everywhere now. Bass, kick drum, snare drum, lead vocal—all the stuff that has a lot of mono correlated information goes a bit to every speaker, except maybe the center. If I put something in the front, I will very rarely put it in the center and the left and the right. I will put it in the center and the surrounds if I want to pull it more into the middle of the room. If I want something off to the side of the room, I'll go left, right, and right surround so it leans to that side."*

—Nathaniel Kunkel

“I like as much information coming out of the surrounds as I do the front, so I’m still as aggressive as I was when I first started, maybe even more so. The only thing that’s really changed for me is how I use the center speaker. I try to use it a little more than I have in the past. I’ve found that when listening in a car, the center speaker is a little more important than it is in the home.”

—Elliot Scheiner

“So I thought, Why don’t I put the vocal in the center monitor most of the time, and the only other things that enter into that monitor are double vocals or harmonies or maybe even a solo instrument? Then I’ll bleed out a little bit of the center vocal into the left and right fronts so if Uncle Bob comes over to the house and sits at that end of the couch, he’s not missing the lead vocal. Then I’ll use divergence and spill a little of that lead vocal into the rear monitors also for that purpose.”

—Greg Penny

What Do I Send To The LFE Channel?

Anything that requires some low-frequency bass extension can be put into the subwoofer via the LFE channel. Many mixers put a little kick and/or bass there if it's used at all.

That said, it might be better not to even use the LFE channel unless you're positive that the subwoofer is calibrated correctly. An uncalibrated subwoofer can cause big surprises in the low end when the track is later played back on the typical home-theater setup. If you don't use the sub when mixing, the low frequencies under 80Hz are naturally folded into the playback subwoofer, resulting in a smooth and even response.

> ***TIP**: The frequency response of the LFE channel only goes up to 120Hz, so you may have to put any instrument that's sent into the LFE into the main channels as well to gain some definition.*

> *"I run the speakers full range so I don't have to put all that much in the LFE. They [consumer electronics manufacturers] all have their own ideas about what bass management is, so I just ignore it and use the LFE to put a minimum of stuff in. Generally, I like to get things to sound how they should sound on my monitors without the LFE."*
>
> *—Elliot Scheiner*

> *"The LFE is a low-frequency effects track. It's used when you have run out of low-frequency headroom in your other channels, so I only go to the .1 when I cannot put any more level on the main channels and I want more bass."*
>
> *—Nathaniel Kunkel*

Mixing Immersive Audio For Virtual Reality

Surround music had pretty much died on the vine when DVD-Audio And SACD discs weren't accepted by the public the way record labels had anticipated. As a result, surround music mixing jobs have been few and far between in the last few years, even though surround for broadcast has become standard operating procedure. That's all changing with the introduction of virtual and augmented reality, where immersive audio (the new term for surround) is a huge part of the experience.

New workstations, plugins and panning tools are being developed at a fast pace, but as of the writing of this book, there's nothing even close to becoming a standard even though a fair number of choices exist. The good thing is that it's cheaper than ever to get into mixing for the format, since a dedicated room or playback system isn't required, as it's all done on headphones!

A couple of tools that are taking hold are Waves NX, which is a virtual monitoring plugin that simulates the ideal acoustics of a mix room, and multichannel panning inside your headphones. Another device that's required is some sort of head tracking device (Waves also offers the NX Head Tracker) that makes the panning follow the movement of your head as you look around the room while wearing a virtual reality headset. Luckily, these products cost less than $100, which makes it easy to experiment with little investment. A new generation of headphones with multiple speakers in order to simulate a surround environment are coming on the market now as well.

As stated before, it's too early in the format's life to provide any true techniques, standards or tools, since it's still a time of experimentation. It's actually a throwback to the early days of 5.1 surround in that regard, but this time, it's a whole lot cheaper to get in the game.

Chapter 7

The Frequency Element: *Using The Equalizer*

Even though an engineer has every intention of making his recording sound as big and as clear as possible during tracking and overdubs, it often happens that the frequency range of some (or even all) of the tracks are somewhat limited when it comes time to mix. This can be due to the tracks being recorded in a different studio where different monitors or signal path was used, the sound of the instruments themselves, or the taste of the artist or producer. When it comes to the mix, it's up to the mixing engineer to extend the frequency range of those tracks if it's appropriate.

In the quest to make things sound bigger, fatter, brighter, and clearer, the equalizer is the chief tool used by most mixers, but perhaps more than any other audio tool, it's how it's used that separates the average engineer from the master.

> *"I tend to like things to sound sort of natural, but I don't care what it takes to make it sound like that. Some people get a very preconceived set of notions that you can't do this or you can't do that, but as Bruce Swedien said to me, he doesn't care if you have to turn the knob around backwards; if it sounds good, it is good. Assuming that you have a reference point that you can trust, of course."*
>
> *—Allen Sides*

> *"I find that the more that I mix, the less I actually EQ, but I'm not afraid to bring up a Pultec and whack it up to +10 if something needs it."*
>
> *—Joe Chiccarelli*

The Goals Of Equalization

While we may not think about it when we're doing it, there are three primary goals when equalizing:

- To make an instrument sound clearer and more defined.
- To make the instrument or mix bigger and larger than life.
- To make all the elements of a mix fit together better by putting each instrument in its own predominate frequency range.

Sometimes just being aware of which of these you're trying to accomplish at the moment can help you get the sound you're looking for quickly and easily, rather than just randomly twisting some knobs until you think it might sound right.

The Frequency Bands and What They Do

Before we examine the various methods of equalization, it's important to note specific areas of the audio frequency bandwidth and how they affect what we hear. The audio band can effectively be broken down into six distinct ranges, each one having an enormous impact on the total sound (see Table 7.1).

Table 7.1: The Audible Frequency Ranges

Range	Description	Effect
16Hz to 60Hz Sub-bass	Encompasses sounds that are often felt more than heard and gives the music a sense of power.	Too much emphasis in this range makes the music sound muddy. Attenuating this range (especially below 40 Hz) can clean up a mix considerably.
60Hz to 250Hz Bass	Contains fundamental notes of the rhythm section.	EQing this range can change the musical balance, making it fat or thin. Too much boost in this range can make the music sound boomy.
250Hz to 2 Hz Low Mids	Contains the low-order harmonics of most musical instruments.	Can introduce a telephone-like quality to the music if boosted too much. Boosting the 500Hz to 1000Hz octave makes the instruments sound horn-like. Boosting the 1kHz to 2kHz octave makes them sound tinny. Excess output in this range can cause listening fatigue.
2kHz to 4kHz High Mids	Contains speech recognition sounds such as "m," "b," and "v."	Too much boost in this range, especially at 3kHz, can introduce a lisping quality to a voice. Too much boost in this range can cause listening fatigue. Dipping the 3kHz range on instrument backgrounds and slightly peaking 3 kHz on vocals can make the vocals audible without having to decrease the instrumental level in mixes where the voice would otherwise seem buried.
4kHz to 6kHz Presence	Responsible for clarity and definition of voices and instruments.	Boosting this range can make the music seem closer to the listener. Reducing the 5kHz content of a mix makes the sound more distant and transparent.
6kHz to 16kHz Brilliance	Controls brilliance and clarity.	Too much emphasis in this range can produce sibilance on the vocals.

di Gar Kulka, Leo. "Equalization - The Highest, Most Sustained Expression of the Recordist's Heart." Recording

For those of you who have an easier time visualizing the audio spectrum in one-octave increments (like those found on a graphic equalizer), here's an octave look at the same chart (see Table 7.2).

Table 7.2: Graphic Equalizer Chart

Octave Band	Effect
31Hz	Rumble, "chest"
63Hz	Bottom
125Hz	Boom, thump, warmth
250Hz	Fullness or mud
500Hz	Honk
1kHz	Whack
2kHz	Crunch
4kHz	Edge
8kHz	Sibilance, definition, "ouch!"
16kHz	Air

EQ Methods

Since each specific song, arrangement, instrument, and player is unique, it's impossible to give anything other than some general guidelines when it comes to equalization methods. That said, there are a number of methods that can quickly and easily get you in the ballpark, as long as you know what you're going for. Remember that different engineers have different ways of arriving at the same end, so if the following doesn't work for you, keep trying. The method doesn't matter, only the end result.

Before these methods are outlined, it's really important that you observe the following:

- **Listen!** Open up your ears and listen carefully to all the nuances of the sound. Everything you hear is important.
- **Make sure you're monitoring at a comfortable level**—not too loud and not too quiet. If it's too quiet, you may be fooled by the non-linearity of the speakers and overcompensate. If it's too loud, certain frequencies may be masked or overemphasized by the non-linearities of the ear itself, and again you will overcompensate.

Method One: Equalize For Definition

Even source material that's been recorded well can sound lifeless, thanks to certain frequencies being overemphasized or others being severely attenuated. More often than not, the lack of definition of an instrument is because of too much lower midrange in approximately the 400 to 800Hz area. This area adds a "boxy" quality to the sound. Sometimes it's because the sound is lacking in the 3kHz to 6kHz area that makes it undefined. Subtractive equalization is a method that allows you to zero in on the frequencies that are masking the definition in a sound.

1. Set the Boost/Cut control to a moderate level of cut (8 or 10dB should work)
2. Sweep through the frequencies until you find the frequency where the sound has the least amount of boxiness and the most definition (see Figure 7.1).
3. Adjust the amount of cut to taste. Be aware that too much cut makes the sound thinner.

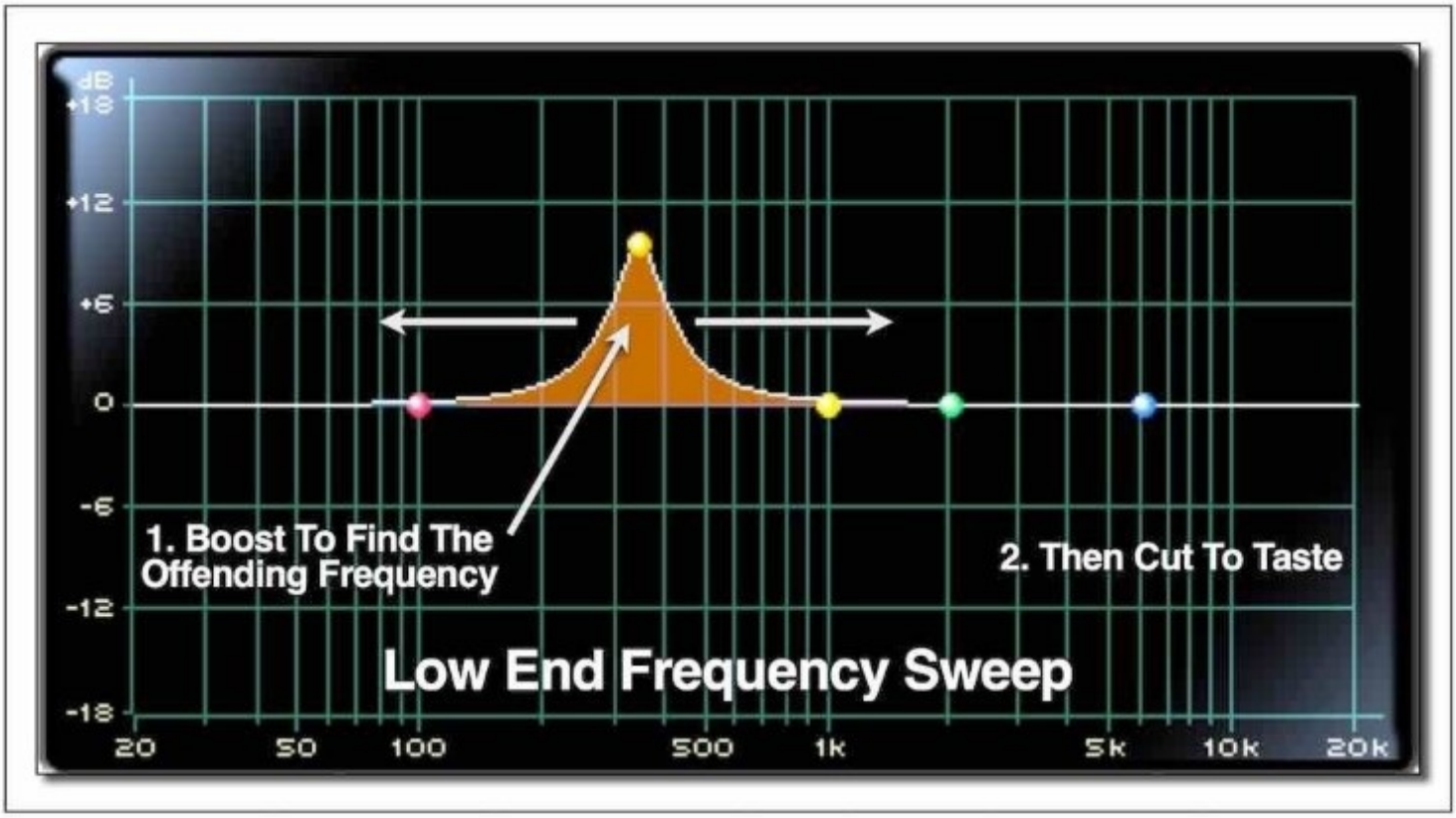

Figure 7.1: Low-end frequency sweep
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There are two spots in the frequency spectrum where the subtractive equalization is particularly effective: between 200Hz and 600Hz and between 2kHz and 4kHz. This is because most directional microphones provide a natural boost at 200 to 600Hz because of the proximity effect brought about by close-miking, and many mics (especially those known for being good vocal mics) have a presence boost between 2kHz and 4kHz. Dipping those frequencies a few dB (more

or less as needed) can make the track sound much more natural than if you were to try to add frequencies instead.

If there were a limited number of microphones (or even just one) used to record all the instruments in a home studio, these two frequency bands (or any other where there's a peak in the response) will build up as more and more instruments are added. By dipping those frequency bands a bit, you'll find that many of the instruments can sit better in the mix without having to add much EQ at all.

> *"What I hate to see is an engineer or producer start EQing before they've heard the sound source. To me, it's kinda like salting and peppering your food before you've tasted it. I always like to listen to the sound source first, whether it's recorded or live, and see how well it holds up without any EQ or whatever."*
>
> *—Bruce Swedien*

***TIP**: Always try attenuating (cutting) the frequency first. This is preferable because all equalizers add phase shift as you boost, which results in an undesirable coloring of the sound. Usually, the more EQ you add, the more phase shift is also added and the harder it may be to fit the instrument into the mix as a result. Many engineers are judicious in their use of EQ, but that being said, anything goes! If it sounds good, it is good.*

Alternate Method

1. Starting with your EQ flat, remove all the bottom end below 100Hz by turning the low-frequency control to full cut.
2. Using the rest of your EQ, tune the mid-upper midrange until the sound is thick yet distinct.
3. Round it out with a supporting lower-mid tone to give it some body.
4. Slowly bring up the mud-inducing bottom end enough to move air, but not so much as to make the sound muddy.
5. Add some high-frequency EQ for definition (see Figure 7.2).

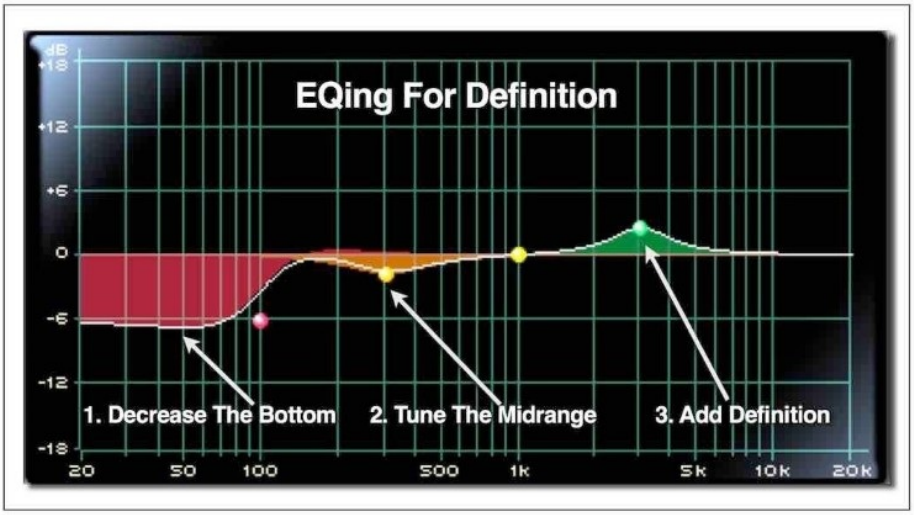

Figure 7.2: EQing for more definition
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> *"I just try to get stuff to sound natural, but at the same time be very vivid. I break it down into roughly three areas: mids, the top and the bottom; then there's low mids and high mids. Generally, except for a very few instruments or a few microphones, cutting flat doesn't sound good to most people's ears, so I'll say, 'Well, if this is a state-of-the-art preamp and a great mic and it doesn't sound that great to me, why?' Well, the midrange is not quite vivid enough. Okay, we'll look at the 3k, 4k range, maybe 2500. Why don't we make it kind of come to life like a shot of cappuccino and open it up a little bit? Then maybe I'm not hearing the air around things, so let's go up to 10k or 15k and just bump it up a little bit and see if we can kind of perk it up. Now, all that sounds good, but our bottom is kind of undefined. We don't have any meat down there. Well, let's sweep through and see what helps the low end. Sometimes, depending on different instruments, a hundred cycles can do wonders for some instruments. Sometimes you need to dip out at 400 cycles, because that's the area that sometimes just clouds up and takes the clarity away,*

but a lot of times adding a little 400 can fatten things up."

—Ed Seay

Method 2: Equalize For Size

Making a sound bigger or larger than life usually comes from the addition of bass and sub-bass frequencies in the 40Hz to 250Hz range, although most will come from an area just below 100Hz, a region just above 100Hz, or both.

To use the method, the low-frequency band of your EQ must be sweepable.

1. Set the Boost/Cut control to a moderate level of Boost (8 or 10dB should work).
2. Sweep through the frequencies in the bass band until you find the frequency where the sound has the desired amount of fullness.
3. Adjust the amount of Boost to taste. Be aware that too much Boost will make the sound muddy.
4. Go to the frequency either half or twice the frequency that you used in Step 2 and add an amount of that frequency as well. Example: If your frequency in Step 2 was 120Hz, go to 60Hz and add a dB or so as well. If your frequency was 50Hz, go to 100Hz and add a bit there (see Figure 7.3).

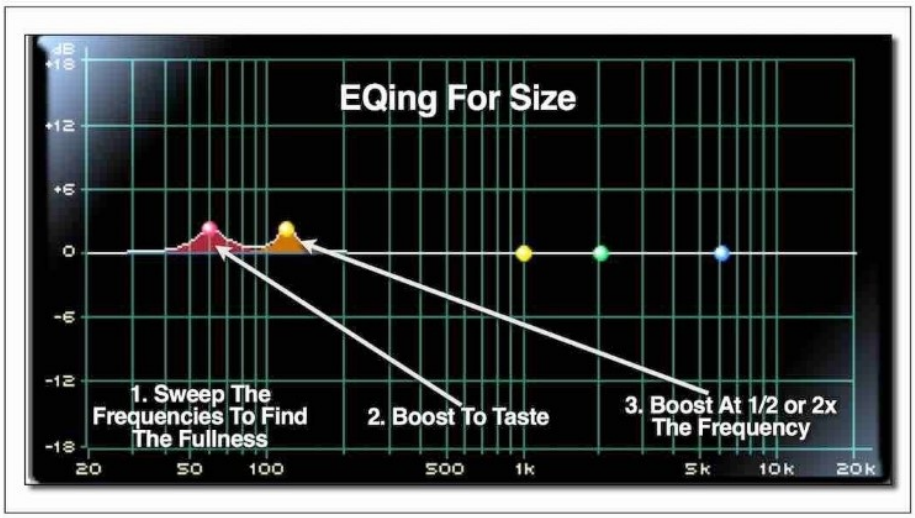

Figure 7.3: EQing for size
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> *"I feel that equalizers are best used when used the least. I use them most to get rid of tones that are somehow not flattering. I'll most often use parametrics, sharp and subtractive, to look for the two or three biggest out-of-sorts characteristics. A snare drum for instance, has any number of boinks that I'll locate, and I may take them out or bring them up as I'm listening to the whole presentation, but I'll already know what and where they are."*

—George Massenburg

Method 3: Juggling Frequencies

Most veteran engineers know that soloing an instrument and equalizing it without hearing the other instruments will probably start making you chase your tail as you make each instrument bigger and brighter sounding. When that happens, you'll find in no time that the instrument you're EQing will begin to conflict with other instruments or vocals frequency-wise. That's why it's important to listen to other instruments while you're EQing. By juggling frequencies, they'll fit together better so that each instrument has its own predominate frequency range. Here's how it's done.

1. Start with the rhythm section (bass and drums). The bass should be clear and distinct when played against the drums, especially the kick and snare. You should be able to hear each instrument distinctly. If not, do the following:
    - Make sure that no two equalizers are boosting at the same frequency. If they are, move one to a slightly higher or lower frequency.
    - If an instrument is cut at a certain frequency, boost the frequency of the other instrument at that same frequency. For example, if the kick is cut at 500Hz, boost the bass at 500Hz (see Figure 7.4).
2. Add the next most predominant element, usually the vocal, and proceed as above.
3. Add the rest of the elements into the mix one by one. As you add each instrument, check it against the previous elements, as above.

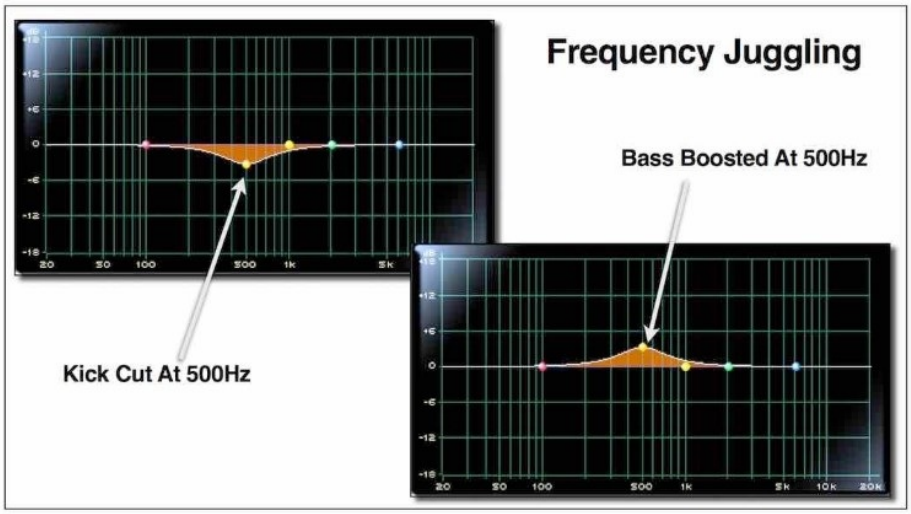

Figure 7.4: Juggling frequencies
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The idea is to hear each instrument clearly, and the best way for that to happen is for each instrument to live in its own frequency band.

***TIP**: It's not uncommon to have to EQ in a circle, where you start with one instrument,*

tweak another that's clashing with it, return to the original one to tweak it, and then go back again over and over until you achieve the desired separation.

> *"I really start searching out the frequencies that are clashing or rubbing against each other, but I really try to keep the whole picture in there most of the time as opposed to really isolating things too much. If there are two or three instruments that are clashing, that's probably where I get more into the solo if I need to kind of hear the whole natural sound of the instrument."*
>
> *—Jon Gass*
>
> *"Frequency juggling is important. You don't EQ everything in the same place. You don't EQ 3k on the vocal and the guitar and the bass and the synth and the piano, because then you have such a buildup there that you have a frequency war going on. Sometimes you can say, 'Well, the piano doesn't need 3k, so let's go lower or let's go higher,' or 'This vocal will pop through if we shine the light not in his nose, but maybe toward his forehead.' In so doing, you can make things audible, and everybody can get some camera time."*
>
> *—Ed Seay*

Easy-To-Remember Golden Rules Of Equalization

- ➡ If it sounds muddy, cut some at 250Hz.
- ➡ If it sounds honky, cut some at 500Hz.
- ➡ Cut if you're trying to make things sound better.
- ➡ Boost if you're trying to make things sound different.
- ➡ You can't boost something that's not there in the first place.

Finding An Offending Frequency

Sometimes a sound has a frequency that sticks out of the mix like a pickaxe in your ear. The method to find it and attenuate it is similar to EQ Method 1.

1. Set the Boost/Cut control to a moderate level of boost. Eight or 10dB should work (see Figure 7.5).
2. Sweep through the frequencies until the frequency that's giving you trouble leaps out.
3. Adjust the amount of cut to taste until the offending frequency is in balance with the rest of the sound. Be aware that too much cut can also decrease the definition of the sound.

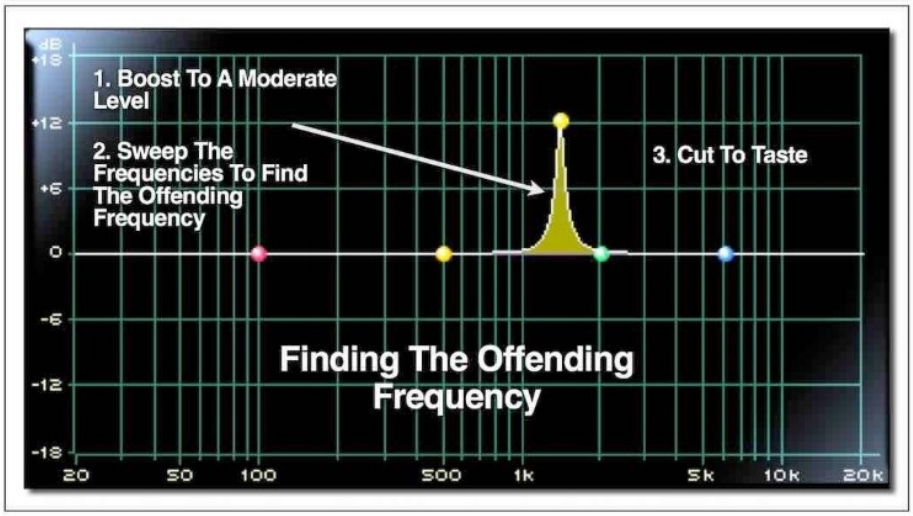

Figure 7.5: Finding the offending frequency

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The Magic High-Pass Filter

One of the most useful and overlooked processors in a mixer's toolkit is the high-pass filter. The low frequencies of many instruments just get in the way of each other and don't add much to the sound. That's why if you roll the low frequencies off below 100 (or even higher) on instruments other than the kick and bass, the mix begins to clean up almost magically (see Figure 7.6). Rolling off the low frequencies of a vocal mic can eliminate the rumble of trucks and machinery that you can't hear anyway because it's so low, yet it muddies up the mix. Even rolling off the bass and drums at between 40Hz and 60Hz can sometimes make the mix a lot louder and punchier without affecting the perceived low end much.

Figure 7.6: The high-pass filter
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***TIP**: After frequency juggling, an instrument might sound terrible when soloed by itself. That's okay. The idea is for it to work in the track.*

> *"It really doesn't matter what it sounds like by itself, because it has to work together with everything else. That's where some of the young producers blow it. They go through and solo tracks and make everything sound fat [by itself]; then when they put it all together, they have a big car wreck.*
>
> *—Jon Gass*

The Magic Frequencies

Every instrument has at least one frequency that might be considered “magic.” That means that you might want to try those frequencies first to make an instrument or voice sound fuller or more distinct. These are outlined in Table 7.3.

Table 7.3: Magic Frequencies For Mix Elements

Instrument	Magic Frequencies
Bass guitar	Bottom at 50 to 80Hz, attack at 700Hz, snap at 2.5kHz
Kick drum	Bottom at 80 to 100Hz, hollowness at 400 Hz, point at 3k to 5kHz
Snare	Fatness at 120 to 240Hz, point at 900Hz, crispness at 5kHz, snap at 10kHz
Rack Toms	Fullness at 240 to 500Hz, attack at 5 to 7kHz
Floor Toms	Fullness at 80Hz, attack at 5kHz
Hi-hat and cymbals	Clang at 200Hz, sparkle at 8k to 10kHz
Electric guitar	Fullness at 240 to 500Hz, presence at 1.5 to 2.5kHz, attenuate at 1kHz for 4 × 12 cabinet sound
Acoustic guitar	Fullness at 80Hz, body at 240Hz, presence at 2k to 5kHz
Organ	Fullness at 80Hz, body at 240Hz, presence at 2 to 5kHz
Piano	Fullness at 80Hz, presence at 3k to 5kHz, honky tonk at 2.5kHz
Horns	Fullness at 120Hz, piercing at 5kHz
Voice	Fullness at 120Hz, boomy at 240Hz, presence at 5kHz, sibilance at 4k to 7kHz, air at 10k to 15kHz
Strings	Fullness at 240Hz, scratchy at 7k to 10kHz
Conga	Ring at 200Hz, slap at 5kHz

6 Trouble Frequency Areas

Whenever an engineer has trouble dialing in the EQ on a track, chances are its because of one or more of the 6 trouble frequency areas. These are areas where too much or too little can cause your track to either stick out like a sore thumb, or disappear into the mix completely. Let's take a look.

- **200Hz (Mud)** - Too much can cause the track or the mix to sound muddy or boomy, while not enough of it can make it sound thin. It's a fine line, but many times mixers err on the side of too much and end up with a track that's too thick that clutters up the mix.

- **300 to 500Hz (Boxy)** - Too much of this frequency area results in the dreaded "boxiness" sound, or if you're listening to a floor tom or kick, the "beach ball" effect. It's also a spot that some less expensive microphones (especially dynamics) tend to emphasize, which is why many mixers almost automatically cut a a few dB of this area out of the kick drum during the mix.

- **800Hz (Walmart)** - Too much in this area results in what's sometimes known as the "Walmart" sound, meaning that it sounds like a cheap stereo purchased in a department store. Try it for yourself - get a cheap pair of computer speakers and you'll find that 800Hz is what you'll mostly hear. Obviously, too much of this frequency range is not a good thing.

- **1k to 1.5kHz (Nasal)** - This is the nasal range of the frequency spectrum and, as the name suggests, too much results in a vocalist that sounds like she's singing through her nose. Once again this is primarily a microphone problem in that it's poorly matched to the vocalist, but notching a bit out during the mix can fix it.

- **4kHz to 6kHz (Presence)** - This frequency range is frequently underutilized during the mix, resulting in a track that lacks definition. Without it, things tend to sound dull, but too much can make the track sound thin or, in the case of a vocal, sibilant.

- **10kHz+ (Air)** - Another widely overlooked frequency band, this provides clarity and adds a certain "realness" to the track. Many vintage mics have a lot of the air frequencies, which is why we prize them for their sound. The Maag Audio EQ4P has a special "Air Band" designed to provide those frequencies with a minimum of phase shift, but you can dial it in other equalizers as well.

Sometimes just tweaking a few of these 6 frequency ranges can take a mix from dull to exciting, or muddy to clear, so keep them in mind during your next mix.

The Relationship Between Bass And Drums

Perhaps the most difficult task of a mixing engineer is balancing the bass and drums (especially the bass and kick). Nothing can make or break a mix faster than how these instruments work together. It's not uncommon for a mixer to spend hours on this balance (both level and frequency) because if the relationship isn't correct, then the song will just never sound big and punchy.

So how do you get this mysterious balance?

To have the impact and punch that most modern mixes exhibit, you have to make a space in your mix for both of these instruments so they won't fight each other and turn the mix into a muddy mess. While simply EQing your bass high and your kick low (or the other way around) might work at its simplest, it's best to have a more in-depth strategy, so to make them fit together, try the following:

1. **EQ the kick drum between 60 and 120Hz, as this will allow it to be heard on smaller speakers.** For more attack and beater click, add between 1kHz and 4kHz. You may also want to dip out some of the boxiness that lives between 200 and 600Hz. EQing in the 30 to 60Hz range will produce a kick that you can feel if your speakers are large enough, but that can also make it sound thin on smaller speakers and probably won't translate well to a variety of speaker systems. Most 22-inch kick drums like to center somewhere around 80Hz, for instance.
2. **Bring up the bass with the kick. The kick and bass should occupy slightly different frequency spaces.** The kick will usually be in the 60 to 80Hz range, whereas the bass will emphasize higher frequencies anywhere from 80 to 250 (although sometimes the two are reversed depending upon the song). Before you continue to EQ at other frequencies, try filtering out any unnecessary bass frequencies (below 30Hz on kick and 50Hz on the bass, although it varies according to style and taste) so the kick and bass are not boomy or muddy. There should be a driving, foundational quality to the combination of these two together.

A common mistake is to emphasize the kick with either too much level or too much EQ and not enough on the bass guitar. This gives you the illusion that your mix is bottom-light, because what you're doing is effectively shortening the duration of the low-frequency envelope in your mix. Since the kick tends to be more transitory than the bass guitar, this gives you the idea that the low-frequency content of your mix is inconsistent. For pop music, it's best to have the kick provide the percussive nature of the bottom while the bass fills out the sustain and musical parts.

3. **Make sure the snare is strong; otherwise, the song will lose its drive when everything else is added in.** This usually calls for at least some compression (see Chapter 9, "The Dynamics Element"). You may need a boost at 1kHz for attack, 120 to 240Hz for fullness,

and 10kHz for snap. As you bring in the other drums and cymbals, you might want to dip a little of 1k on these to make room for the snare. Also, make sure that the toms aren't too boomy. (If so, try rolling them off a bit below 60Hz first before you begin to EQ elsewhere.)

4. **If you're having trouble with the mix because it's sounding cloudy and muddy on the bottom end, turn off the kick drum and bass to determine what else might be in the way in the low end.** You might not realize that there are some frequencies in the mix that aren't musically necessary. With piano or guitar, you're mainly looking for the mids and top end to cut through, while any low end might be just getting in the way of the kick and bass, so it's best to clear some of that out with a high-pass filter. When soloed the instrument might sound too thin, but with the rest of the mix the bass will sound so much better, and you won't really be missing that low end from the other instruments. Now the mix will sound louder, clearer, and fuller. Be careful not to cut too much low end from the other instruments, as you might lose the warmth of the mix.
5. **For dance music, be aware of kick drum to bass melody dissonance.** The bass line is very important and needs to work very well with the kick drum when it's reproduced over the huge sound systems commonly found in today's clubs. If your kick has a center frequency of an A note and the bass line is tuned to A#, they're going to clash. Tune your kick samples to the bass lines (or vice versa) where needed.
6. **If you feel that you don't have enough bass or kick, boost the level, not the EQ.** This is a mistake that everyone makes when they're first getting their mixing chops together. Most bass drums and bass guitars have plenty of low end and don't need much more, so be sure that their level together and with the rest of the mix is correct before you go adding EQ. Even then, a little goes a long way.

> *"I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom. Sometimes you can have a really thin kick by itself, but when you put the bass with it, it seems to have enough bottom because the bass has more bottom end. I build the drums on top of that."*
>
> *—Benny Faccone*

> *"I'll get the drums happening to where they have some ambience, then put the vocal up and get that to where that's sitting right. Then I'll start with the bass and make sure that the kick and the bass are occupying their own territory and not fighting each other."*
>
> *—Jerry Finn*

EQ Techniques

General Tips

- Use a narrow Q (bandwidth) when cutting and a wide Q when boosting.
- If you want a sound to stick out of the mix, roll off the bottom; if you want it to blend in, roll off the top.
- The fewer instruments that are in the mix, the bigger each one should sound.
- Conversely, the more instruments in the mix, the smaller each one needs to be for everything to fit together.
- It’s usually better to add a small amount at two frequencies than a large amount at one.
- Be aware that making an instrument sound great while soloed may make it impossible to fit together with other instruments in the mix.

For Snare

To find the “point” on the snare, boost the upper midrange about 5 or 6dB at 2kHz or so. Open up the bandwidth (if that parameter is available) until you get the snare to jump out, then tighten the bandwidth until you get only the part of the snare sound that you want most. Then fine-tune the frequency until you need the least amount of boost to make it sit in the mix yet have some definition.

For Drums

“In the old days you always pulled out a little 400 on the kick drum. You always added a little 3k and 6k to the toms. That just doesn’t happen as much anymore because when I get the project, even with live bands, the producer’s already triggered the sound he wanted off the live performance, so the drums are closer.”

—Dave Pensado

For Kick

- Try boosting 4kHz, cut 200 to 400Hz, and then boost the low frequencies around 60 to 100Hz to find the drum's resonance.
- One fairly common effect used in R&B is to trigger a 32Hz tone with the kick, then gate it so it has the same volume envelope. Blend and compress both the original kick and the 32Hz tone to taste.
- For a metal kick, add a bit of 3kHz or so on the kick drum to function as the "nail in the paddle"sound.
- For a kick meant for a club, emphasize the 200 to 300Hz range while rolling off the extreme low end. The club system makes up the difference, so if you mix the bottom of it the way you think you'll hear it in a club, you're probably going to overload the house sound system.
- If your bass is a very pure sine wave–like sound and your kick is an 808, they may mask each other. If the kick is lower-sounding than the bass, add a sample with some mid or top punch. If the kick is higher than the bass, you can add some distortion or a plugins like MaxxBass (see Figure 7.7) to add higher harmonics to the bass. Make sure you check both on small speakers.

Figure 7.7: Waves MaxxBass
Courtesy of Waves Audio Ltd

For Bass

- The ratio between the low bass (80 to 120Hz) and the mid bass (120 to 200Hz) is important. Try using two fairly narrow-peaking bands, one at 100Hz and another at 150Hz, boosting one and cutting the other slightly. If the bass is too "warm," sometimes reducing the lower band can make it more distinct without removing the deeper fundamentals that live in the 100Hz band.

- For clarity on the bass, try boosting some of the 800Hz area, since this can provide definition without getting too snappy-sounding.

- A four-band parametric will allow you to adjust several bands below 200Hz. Try attenuating the low frequencies around 40 to 70Hz, then slightly boosting the frequencies from 80 to 120Hz where the fundamental lies, then boost the frequencies from 130 to 200Hz where the over- tones and cabinet/neck/body resonances live.

- Muddy bass can mean a lot of things, but at a minimum it usually involves a lack of presence of the higher harmonics. Most bass tracks have a sweet spot between 60Hz and 1.2kHz where the upper-order harmonics sing, and this is the place to boost for more presence in the mix.

- Take a low-cut filter and center it at 250Hz so that all the lows of the bass are attenuated. Now take a bell-shaped EQ and boost it 4dB with a narrow band and sweep around the 80 to 180Hz region to find where your bass frequencies fit best in the track. Once you find it, widen the bandwidth and boost more if necessary. If you want more density on the bottom, you may need to do this with another bell filter on the frequencies below the previous one. This should tighten up the low end, add space for a kick drum, and make your mix less boomy.

- High-pass filter the bass anywhere from 40 to 80Hz. It's amazing how much that can help the bass tighten up sometimes.

- Any instrument with low-frequency content (below 500Hz) can affect the sound of the bass. This includes kick, keyboards, (male) vocals, double bass, celli, low-tuned guitars, and so on. Cut the low end (anywhere below 80 to120Hz) from tracks where the low end isn't needed. This will help those tracks to cut through while leaving more space in the mix for the bass and the kick.

- Removing the 250Hz region information from instruments such as guitars, keyboards, and even vocals is often of more use than cutting it from the bass.

- To achieve more definition from a bass guitar, first make a duplicate of the bass track and then process the duplicate track in your DAW with the Moogerfooger Low-Pass plugins. Set the plugins's Mix parameter to 10, the Resonance to 1, and the overall

Amount to 1. Now turn the frequency control up until you get a well-defined sub tone. Group both bass tracks to a separate bus and create a new aux track. Be sure to assign the input to the same bus. Then, on the original bass track in your DAW, use an EQ plugins to roll some of the low subs out. Next, blend the original bass track with the Moogerfooger track to create a fat, solid composite bass sound. The aux track will now become the bass master track. Finally, EQ and compress that aux track to fine-tune the bass sound to taste. The same tip can also be used on kick drums.

- With hip-hop and electronic music, the bass tends to contain a lot of information in the 30 to 60Hz range that you can feel. Many hip-hop or EDM records will raise the low-frequency target area slightly higher to the 70 to 100Hz range and elongate the duration to create the illusion that there's a lot of bass information so that it can sound full on smaller monitors. Be careful not to over-EQ, though. Clubs and cars with huge bass drivers are already hyped in this frequency range.

- With rock bass, the idea is to create an aggressive in-your-face bass sound. For this the focus will be mainly on the amp sound. Boost anywhere between 50 and 100Hz for the bottom end, dip between 400 to 800Hz (this will allow the guitars and vocal to have more room to speak musically) and boost between 1.5 and 2.5kHz for midrange.

***TIP**: Be aware that mixing the DI sound with the amp sound might cause phasing problems in the midrange, so be sure to check the polarity with the phase parameter and use the selection with the most bass. (Although sometimes the out-of-phase sound can work better in the mix, so be sure to check.)*

> *"The one thing that did change for me over time was my not liking 200Hz. That frequency couldn't be touched in the early days because we didn't have an EQ that was centered there, and it wasn't until later on that I decided that I didn't like it and began to pull it out [of the bass and kick."*
>
> *—Ken Scott*

For Guitars

- For a fatter-sounding guitar, boost the midrange about 9dB or so and sweep the frequencies until you hear the range where the guitar sounds thick but still bright enough to cut through the mix. Back the boost down to about the point where the guitar cuts through the mix without being too full-sounding.

> *"I use EQ differently from some people. I don't just use it to brighten or fatten something up, I use it to make an instrument feel better. Like on a guitar, instead of just brightening up the high strings and adding mud to the low strings, I may look for a certain chord to hear more of the A string. If the D string is missing in a chord, I like to EQ and boost it way up to +8 or +10 and then just dial through the different frequencies until I hear what they're doing to the guitar. I'm trying to make things more balanced in the way they lay with other instruments."*
>
> *—Don Smith*

- Boosting 10kHz on guitars will accentuate finger noise and tiny string movements. Boosting 5 to 8kHz will allow the guitar to better cut through the mix. Boosting 1k to 5kHz will give the guitar more presence.

- Consider filtering the guitar with both high-pass and low-pass filters. If you leave too much low end in a distorted electric guitar, it will compete with the rhythm section. Too much above 8kHz can compete with the cymbals.

For Vocals

- Boost a little 125 to 250Hz to accentuate the voice fundamental and make it more “chesty” sounding.
- 2k to 4kHz accentuates the consonants and makes the vocal seem closer to the listener.

> *“On a vocal sometimes I think, ‘Does this vocal need a diet plan? Does he need to lose some flab down there? Or sometimes we need some weight on this guy, so let’s add some 300 cycles and make him sound a little more important.’”*
>
> *—Ed Seay*

> *“I think of EQ as an effect, much the same way you would add chorus or reverb to a particular instrument or vocal. For example, I might have a vocal where I think it’s really EQed nicely, and then I’ll add a little more 3k just to get it to bite a little more. Then it just makes me feel like the singer was trying harder, and it brings out a little bit of passion in his or her voice.”*
>
> *—Dave Pensado*

Chapter 8

The Dimension Element: *Adding Effects*

The fourth element of a mix is dimension, which is the ambient field that the track or tracks sits in. Dimension can be captured while recording, but it's usually created or enhanced when mixing by adding effects such as reverb, delay, or modulation. Dimension might mean re-creating an acoustic environment, but it could also mean adding width or depth to a track to try to spruce up a boring sound. There are four reasons why a mixer might add dimension to a track:

- To create an aural space
- To add excitement
- To make a track sound bigger, wider, or deeper
- To move a track back in the mix (giving the impression that it's farther away)

> *"Everything has to be bigger always. Effects are makeup. It's cosmetic surgery. I can take a very great song by a very great band and mix it with no effects on it at all, and it'll sound good, and I can take the same song and mix it with effects and it'll sound fantastic!"*
>
> *—Lee DeCarlo*

> *"The way I think of it is the pan knob places you left to right while the effects tend to place you front to rear. In other words, if you want the singer to sound like she's standing behind the snare drum, leave the snare drum dry and wet down the singer, and it'll sound like the singer is standing that far behind the snare drum. If you want the singer in front of the snare drum, leave him dry and wet down the snare drum."*
>
> *—Dave Pensado*
>
> *"Sometimes [I add effects for] depth, and sometimes you just want it to sound a little bit more glamorous. I've done records where I didn't use any effects or any verb, but quite often just a little can make a difference. You don't even have to hear it, but you can sense it when it goes away. Obviously, an effect is an ear-catcher or something that can just kind of slap somebody and wake them up a little bit in case they're dozing off there."*
>
> *—Ed Seay*

One of the reasons why we record elements in stereo is to capture the natural ambience (or

dimension) of an instrument. Because we can’t always record everything this way, we must create this aural space artificially.

The Six Principles For Adding Effects

There are six principles that offer a general guideline on how effects can be used.

Principle 1: Picture the performer in an acoustic space and then re-create that space around him.

This method usually saves time, as it's a lot faster than simply experimenting with different effects presets until something excites you (although that method can work, too). Also, the artificially created acoustic space needn't be a natural one. In fact, as long as it fits the music, the more creative the better.

Principle 2: Smaller reverbs or short delays make things sound bigger.

Reverbs with decays under a second (and usually much shorter than that) and delays under 100 milliseconds (again, usually a lot shorter than that) tend to make the track sound bigger rather than push it back in the mix, especially if the reverb or delay is in stereo.

Many times a reverb will be used with the decay parameter turned down as far as it will go (which may be as low as 0.1 seconds), but this setting is sometimes the most difficult for a digital reverb to reproduce, resulting in a metallic sound. If this occurs, sometimes lengthening the decay time a little or trying a different preset will result in a smoother, less tinny sound, or you can try another plugin or hardware unit that performs better under these conditions.

Principle 3: Long delays, long reverb predelay, and reverb decay settings push a sound further away if the level of the effect is loud enough.

Delays and predelays (see the section later in the chapter on reverb) longer than 100 milliseconds are distinctly heard and begin to push the sound away from the listener.

The trick between something sounding big or just distant is the level of the effect. When the decay or delay is short and the level is loud, the track sounds big. When the decay or delay is long and loud, the track just sounds far away.

Principle 4: If delays are timed to the tempo of the track, they add depth without being noticeable.

Most mixers set the delay time to the tempo of the track. (See the section on delay for how to do this.) This makes the delay pulse with the music and adds a reverb type of environment to the sound. It also makes the delay seem to disappear as a discrete repeat, but it still adds a smoothing quality to the mix element.

Principle 5: If delays are not timed to the tempo of the track, they stick out.

Sometimes you want to distinctly hear a delay and the best way to do that is to make sure that the delay is not exactly timed to the track. Start by first putting the delay in time; then slowly alter the

timing until the desired effect is achieved.

Principle 6: Reverbs sound smoother when timed to the tempo of the track.
Reverbs are timed to the track by triggering them off of a snare hit and adjusting the decay parameter so that the decay just dies before another snare hit. The idea is to make the decay "breathe" with the track. One way to achieve this is by making everything as big as possible at the shortest setting first, then gradually making the settings longer until it's in time with the track.

Of course, the biggest part of adding effects to a mix is experience, but keeping these principles in mind will provide a perfect place to start.

Using Delays

Delays are a secret weapon of many mixers who sometimes opt to use several of them instead of reverb to add depth to the mix. If set up correctly, a delay can both pulse with and blend into a track, making it seem both deeper and fatter without calling attention to itself.

Types Of Delays

There are a number of different delays commonly used in a mix. Let's look at them.

- **Haas effect.** This is a delay of 40 milliseconds or less that's not perceived as a separate repeat but can add a sense of spaciousness if panned opposite the source.
- **Short**. A short delay is generally anywhere from 40 milliseconds to around 150. The idea is to add a double- tracked effect. This can be heard on old Elvis records as well as others from that era.
- **Medium**. A medium delay is anywhere from 150 milliseconds to around 400. Even though we hear it as a distinct repeat, this length of delay is more for adding a sense of space around the source, even though it may be somewhat imperceptible sometimes if timed to the track.
- **Long**. A long delay is anywhere from 400 milliseconds up to about a second (1,000 milliseconds). You hear a long delay as a very distinct and specific repeat.
- **Stereo**. A stereo delay allows for a different delay time on each side of the stereo soundfield.
- **Ping-pong.** A delay that bounces from one side of the stereo image to the other.
- **Tape**. In the analog days, delay was accomplished by using an outboard tape machine. The delay occurred because the playback head was located after the record head, which created a time delay (see Figure 8.1). As the speed of the tape machine changed, so would the delay. For example, a 15-IPS tape speed would result in a delay somewhere in the 125 to 175 milliseconds range. (It would be different with different models of tape machines because the gap between the heads was different for each.) At 7 1/2 IPS the delay would double to around 250 to 350 milliseconds. Because of the analog nature of magnetic tape, it has the characteristics of wow and flutter of the tape path, plus a rolled-off high-frequency response and increased distortion with each repeat, which most tape delay plugins try to emulate.

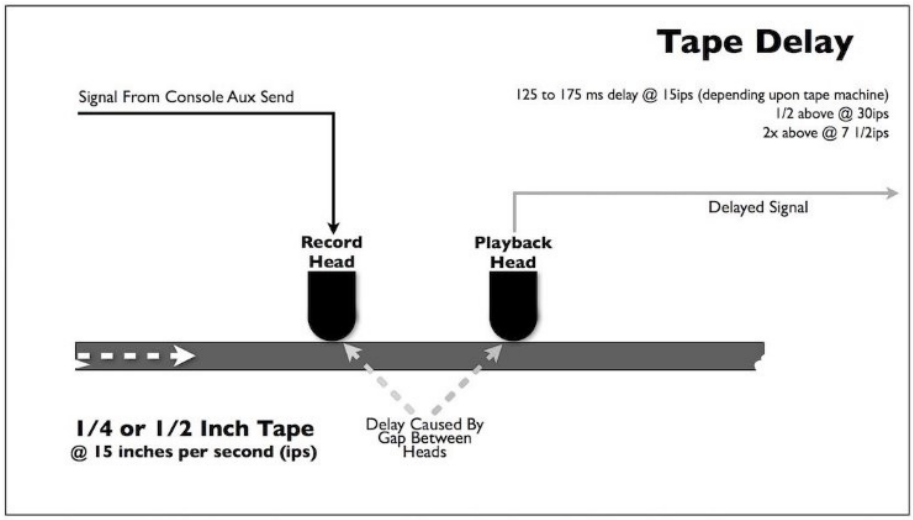

Figure 8.1: Delay using a tape machine
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Timing Delays To The Track

It's important to time the delay to the track in order for it to pulse with the song. When this happens, the delay can almost seem to disappear from the mix, since you might not hear distinct repeats, but it can add a depth or glue to the track that can't be achieved any other way.

Determining The Song's Tempo

Before we can time the delay to a track, we first have to determine the tempo of the track. Sometimes the tempo is predetermined in your DAW when the track is created, while many delays allow you to tap it in. Alternatively, you can use an external tool such as the Delay Genie iPhone app. If you want to determine the beats per minute (bpm) the old-fashioned way, here's how it's done.

The Classic Way Of Calculating The Tempo Of A Song

1. Count the number of beats in a song for 15 seconds.
2. Multiply the number of beats by 4. For example, 24 beats × 4 = 96 bpm

Calculating The Delay Time

Delays are measured tempo-wise using musical notes in relation to the tempo of the track. In other words, if the song has a tempo of 120 bpm, then the length of time it takes a quarter note to play would be 1/2 second (60 seconds ÷ 120 bpm = 0.5 seconds). Therefore, a quarter-note delay would be 0.5 seconds or 500 milliseconds (0.5 × 1000ms per second), which is how almost all delay devices are calibrated.

But 500ms might be too long and set the source track too far back in the mix. To get smaller delay increments, do the following:

1. Divide a 1/4-note delay in half for an 1/8-note delay (500ms ÷ 2 = 250ms).

2. 2. Divide it in half again for a 1/16-note delay (250ms ÷ 2 = 125ms).
3. Divide it in half again for a 1/32-note delay (125 ÷ 2 = 62.5ms, or rounded up to 63).
4. That still might not be short enough for you, so divide it in half again for 1/64 note (62.5 ÷ 2 = 31.25, rounded off to 31ms).
5. Again, this might not be short enough, so divide it in half again for 1/128 note (31ms ÷ 2 = 15.625, rounded up to 16ms).
6. And yet this still might not be short enough, so divide again for 1/256 note (16ms ÷ 2 = 8ms).

Now, such small increments like 8 and 16ms might not seem like much, but they're used all the time to make a sound bigger and wider. Even a short delay like this will fit much more smoothly into the track if it's timed.

> *"I use a lot of 10, 12, 15ms on things. In the R&B stuff, you get a lot of stereo tracks that really aren't stereo. One of the first things I do is to widen the thing out, even if it's only 3, 5, or 10 milliseconds, and just get that stuff separated so I can keep my center cleared out."*
>
> *—Jon Gass*

The Classic Way Of Calculating the Delay Time

1. Start a stopwatch when the song is playing and count 25 beats
2. Stop the stopwatch on the 25th beat and multiply the time by 41.81

The result is the delay time in milliseconds for a quarter-note delay

or
60,000 ÷ song tempo (in beats per minute)
The result equals the delay time in milliseconds for a quarter-note delay

It's also possible (and sometimes even preferable) to use other note denominations, such as triplets or dotted eights, sixteenths, and so on. These odd note denominations can be determined by using the following formula:

Delay Time × 1.5 = Dotted Value

Example: 500ms (quarter-note 120 bpm delay) × 1.5 = 750ms (dotted quarter note)

Delay Time × 0.667 = Triplet Value

Example: 500ms (quarter-note 120 bpm delay) × 0.667 = 333.5ms (quarter-note triplet)
As with the straight notes (quarter, eighth, and so on), you can continually divide the above values in half until you get the desired denomination.

> ***TIP**: Triplet or dotted-note timed delays sometimes feel better in a track and give it more "glue" because of the way they blend with the pulse of the track.*

Setting The Repeats

The number of repeats is selected by the Feedback control (sometimes called Regeneration), which sends some of the output of the delay back into the input. The amount is usually set in percentages, with 0% resulting in a single repeat and 100% resulting in an infinite loop of repeats that eventually becomes feedback that gets louder and louder.

Although the number of repeats is dependent upon the song and its tempo, most of the time delays are set up for one to three repeats (less than 20% feedback). Too many repeats can sometimes result in the track sounding muddy.

The best way to set the repeats is to only have enough to either fill a hole in the arrangement or fill the space until the next event. In the case of a vocal, the repeats would occur from the end of a phrase only until the next phrase began. Many delay plugins have a feature called *dynamic delay*, where the delay only begins after the source has died away and then automatically ceases the repeats when the next phrase begins again. This automatically keeps the delay out of the way, but the benefits of having a delay on all the time are negated.

Typical Delay Setups

Many mixers use standard delay setups of short and long; short, medium, and long; or even full note, triplet, and dotted note. Many times the delays will be set up during the session prep, but the timing will be determined during the mix.

Here's an example of a two-delay short and long setup, where if the tempo of the song is 105 bpm, the short delay is set to 1/32 note of 71 milliseconds and the long delay is set to a dotted 1/16 note of 214 milliseconds (see Figure 8.2).

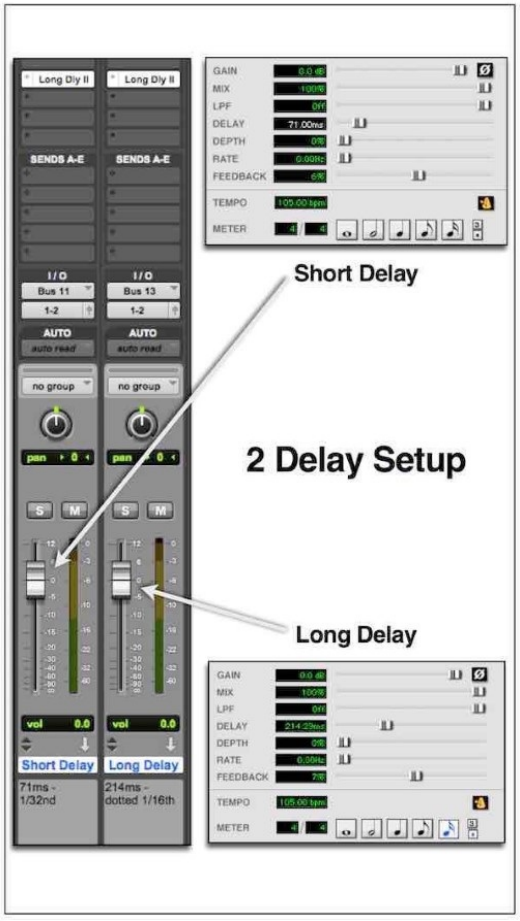

Figure 8.2: A two-delay short and long setup
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In an example of a three-delay setup at a song tempo of 105 bpm, the short delay might be set to a 1/32-note triplet at 48 milliseconds, the medium delay to a 1/16 note at 143 milliseconds, and the long delay to a 1/4-note triplet at 381 milliseconds (see Figure 8.3).

Figure 8.3: A three-delay setup
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Of course these are only examples, and the real note selections and times are strictly based on how they feel in the song. It usually takes some experimentation to hear exactly what fits and what doesn't.

Delay Techniques

Here are a number of techniques often used for particular mix elements. Don't limit yourself to the examples cited, though, as they can easily work for other instruments, vocals, or program sources as well.

For Vocals:

- A stereo delay with a 1/4 or ⅛ note delay on one side and a 1/4 or ⅛ note triplet or dotted note on the other provides movement along with depth and is a favorite trick of EDM mixers.

- To simulate a vocal double, dial in a 1/16 note delay and then modulate it (see the "Using Modulation" section of this chapter) so it slowly raises and lowers in pitch. If the modulation can be set so it's random, it will sound more realistic.

- For a quick vocal effect to give it some space and depth during tracking or overdubs, set up a mono 220ms delay with a couple of repeats.

- Paul McCartney reportedly uses a 175ms delay on his vocals almost all the time.

- For getting a dry vocal to jump out, use two bandwidth-limited (at about 400Hz to 2.5kHz) delays in the neighborhood of 12ms to the left and 14ms to the right, each panned slightly off center. Bring up the delays until you can hear them in the mix and then back it off to where you can't. Occasionally mute the returns to make sure it's still bringing the vocals out as they sit well into the rest of the balance. You can also time the delays to a 1/64 note on one side and a 1/128 note on the other.

> *"I like a vocal mostly dry, but then it usually doesn't sound big enough. You want the vocalist to sound like they're really powerful and dynamic and just giving it everything, so I'll put an 1/8-note delay on the vocal but subtract a 1/16, a 1/32, or a 1/64 note value from that 1/8 note. What it does is give a movement to the delay and makes the singer have an urgency that's kind of neat. I put the 1/8 minus 1/64 on the left side and put the straight 1/8 note on the right side. You can experiment with pushing the pitch up a little bit on one side and down on another too if your singer's a little pitchy, since that usually makes them sound a bit more in tune. Sometimes putting the ⅛ note triplet on one side and the straight 1/8 note on the other, if you've got any kind of swing elements of the track, will make the vocal big, yet it doesn't make the singer sound like he's taking a step back."*
>
> *—Dave Pensado*

- During the ‘80s, when guitars were often recorded direct, many LA session guitarists used a short stereo delay of 25ms on one side and 50ms on the other to provide some space around the sound.

For Guitars:

- To make the guitar sound larger than life, set a delay at less than 100ms (timed if you can) and pan the guitar to one side and the delay to the other.
- Use a mono delay on the guitar set to about 12ms (or whatever the tempo dictates) and hard-pan both the guitar and the delay. This makes the guitar sound much bigger and almost like two people playing perfectly in sync, yet still keeps a nice hole open in the middle for the vocals.
- Pan the guitar track and the delay to the center (or put your monitors in mono); then slowly increase the delay time until it sounds bigger. Increase it a little more for good measure. You’ll probably find the result is in the area of 25 to 30ms.

***TIP**: Instead of always syncing your delay to the tempo of the song in your DAW, try tapping the tempo manually instead or set the delay slightly ahead or behind the beat for a little more natural groove.*

For Keyboards:

- A stereo delay setting of 211ms on one side and 222 on the other provides a quick and easy room simulation and adds some life to a directly recorded keyboard.

Using Reverb

Reverb is one of the two principle ways that we can add artificial ambience to a track in a mix, but there's a lot more to it than just dialing up a preset. While sometimes that may work, more often than not, a reverb's settings are as carefully crafted as the mix itself.

Types Of Reverb

There are six primary categories of reverb, all with a different sonic character; three of these are actual acoustic spaces, one is an analog way to reproduce one, and one is not found in nature but can really sound cool. The reason why there's a difference is that just like everything else in music and audio, there are many paths to the same end result. You'll find that every digital reverb plugin or hardware unit provides its own version of these sounds.

- **Hall**. A hall is a large space that has a long decay time and lots of reflections. Sometimes there's a subcategory of the hall reverb called "church," which is just a more reflective hall with a longer decay.
- **Room**. A room is a much smaller space that can be dead or reflective, depending upon the material that the walls, floor, and ceiling are made of. It usually has a short decay time of about 1.5 seconds or less.
- **Chamber**. An acoustic chamber is a dedicated tiled room that many large studios used to build to create reverb (see Figure 8.4). Phil Spector's "Wall of Sound" was built around an excellent acoustic chamber at Gold Star Studios in Hollywood (long-since closed, unfortunately), for example. The acoustic chambers at Capitol Studios, designed by Les Paul himself, still have a reverb sound that is revered by mixers everywhere. Other common artificial spaces used as acoustic chambers include showers and stairwells.

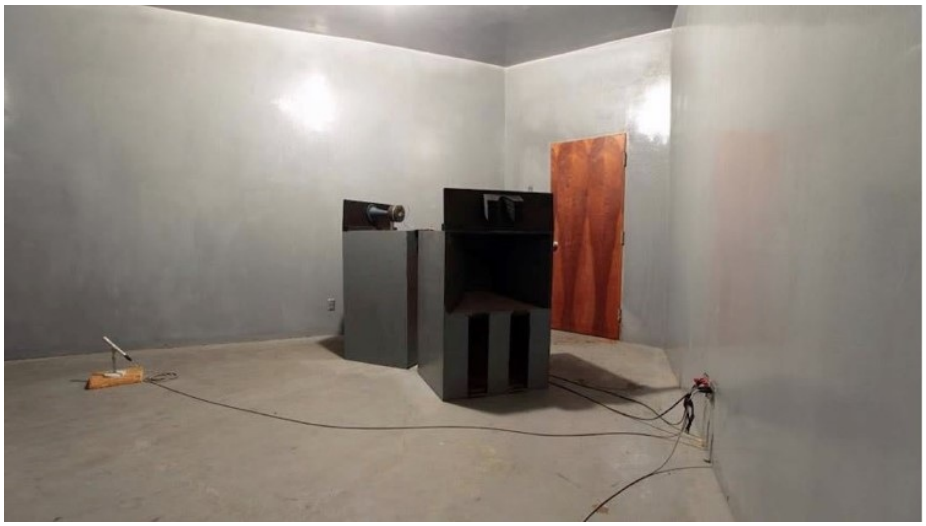

Figure 8.4: An acoustic chamber
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- **Plate**. A plate is a 4 foot by 6 foot hanging piece of sheet metal with transducers attached to it that many studios used for artificial reverb when they couldn't afford to build a chamber (see Figure 8.5). The first plate reverb was the EMT 140, developed

in the late 1950s, and is still held in high esteem by many mixers for its smooth sound.

Figure 8.5: A plate reverb

- **Spring**. A spring reverb is just that; a number of springs that delay the signal at different times to create reverb. Spring reverbs have always been an inexpensive, although limited way to create the effect. Some where made inexpensive enough to be included on early Fender guitar amps of the 60s, a big innovation at the time. Spring reverbs have generally been a lot less flexible in terms of parameter control, although some of the larger units intended for the studio (like the AKG BX15 - see Figure 8.6) provided more control than most of the least expensive versions. Today, there are excellent software versions of spring reverbs available by numerous manufacturers that add far more parameter control than their hardware counterparts.

Figure 8.6: A spring reverb

- **Non-linear.** The non-linear category is strictly a product of modern digital reverbs, as

the sound isn't found anywhere in nature. While natural reverbs decay in a rather smooth manner, once a reverb is created digitally, it's possible to make that decay happen in unusual ways. The reverb tail can be reversed so it builds instead of decays, or it can be made to decay abruptly, both of which makes the decay "non-linear." This preset was a popular mixing effect used on drums during the '80s, when the feature first became available on the AMS RMX 16 digital reverb (see Figure 8.7).

Figure 8.7: The AMS RMX 16 digital reverb

As to which reverb category to use, that's strictly up to the taste of the mixer and how he sees it fitting with the song. Many mixers might always use a room or a chamber on drums, a plate on vocals or guitars, and a hall on strings or keyboards, while others may do just the opposite. Many might refine the sounds of each and finally settle on a few that they feel always work in a particular situation with a certain instrument or vocal. It's all up to experience.

All of these reverbs can be modeled using what's known as a convolution reverb that uses a quick burst of audio energy (called an impulse) to excite the room or device, which then allows it to sample its parameters. Examples of convolution reverbs include the Audio Ease Altiverb and Avid's TL Space.

Timing Reverbs To The Track

Just like delays, reverbs sound smoother if they're timed to the pulse of the track. Doing this adds depth without sticking out and makes the mix seem more polished. The two parameters that are adjusted for timing are the decay time and the predelay.

Timing The Decay

In simple terms, the decay time is the time it takes for the reverb tail to die out. If the decay time is timed to the pulse or bpm of the song, the track seems tighter and cleaner while still retaining all of the depth.

To time the decay time to the track, trigger the reverb with the snare and adjust the decay parameter so that the decay just dies by either the next snare hit or a later one. The idea is to make the decay "breathe" with the track. You can use this decay time for the other reverbs, but you'll probably have to adjust them slightly because the decay response of every reverb or reverb setting (such as hall, plate, chamber, room) is different due the characteristics of the reverb itself.

> ***TIP**: If the decay sounds too short after just one hit, time it so the decay dies by the end of the second or even the fourth snare hit.*

Timing The Predelay

Predelay means delaying the reverb entrance slightly after you hear the source signal. The reason it's used is so the source signal doesn't sound washed out in ambience. With a little bit of predelay, you'll hear the source's attack, then the reverb, so the source signal has more definition as a result.

Predelay is usually timed to the tempo of the track. Back in the days of real plates and chambers, predelay was achieved by using the slap delay from a tape machine, but today it's a standard parameter on every reverb plugin or hardware device.

The same way that you determined the delay time for the track provides the timing for the predelay. The difference is that you usually need a smaller increment than you might've used for a delay, and it's usually less than 100 milliseconds.

For instance, if you determined that a suitable 1/16-note delay time is 150ms, cut it in half (75ms), then cut it in half again (37.5ms), and maybe even in half again (19ms, rounding it off). That's probably going to be the best timing to start with, but don't be afraid to try the longer or shorter variations as well.

> ***TIP**: A predelay in the 20 to 40ms range is the most common. If you don't want to time it, just start with 20ms as a good compromise.*

Of these two parameters, the predelay is probably the most important in that the reverb seems more a part of the track when that parameter is timed. If you really want the reverb to stick out a bit, just randomly select a predelay time or use none at all.

Typical Reverb Setups

Just like with delays, many mixers have different combinations of reverbs that they use. While some old-school engineers might only use a single reverb on a mix and it will sound great, some will use many more as a matter of course.

In many cases, two reverbs are used, with one set to short decay and used on drums and the second with a longer decay and used on the other mix elements. Figure 8.8 shows a typical two-reverb setup with a parameters that will be timed to the pulse of the song (which, like the example for the delays, is 105 bpm). The short reverb is set to a room reverb with an 18ms predelay (1/128 note) and a 1.2-second decay, while the long reverb is set to a plate sound with a 36ms predelay and a 1.8-second decay time.

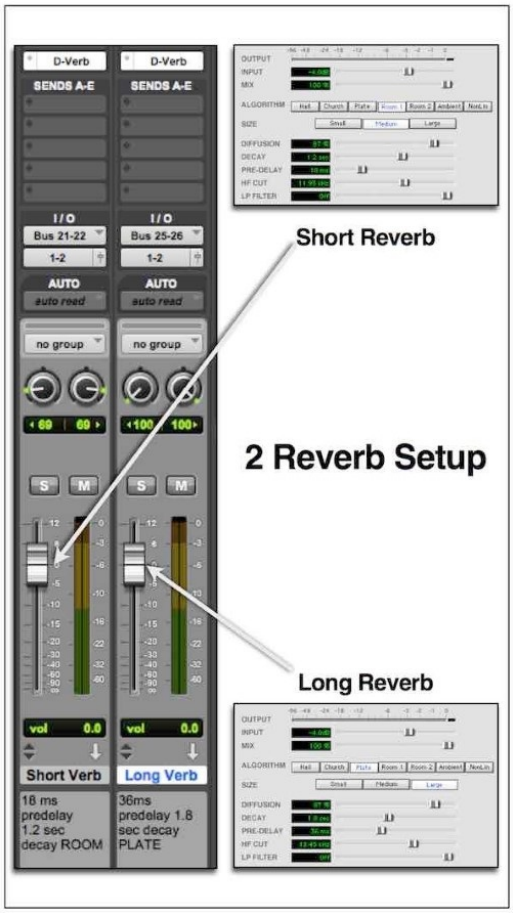

Figure 8.8: A two-reverb setup
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> ***TIP**: The decay time can never accurately be predicted by the bpm of the song since the rate of decay is different for each model of reverb.*

In a three-reverb setup, we may use different reverb categories, but the decay is relegated to short, medium, and long. In Figure 8.9 the short reverb is set to a plate with a decay time of .8 seconds, the medium reverb is a chamber set to a decay time of 1.4 seconds, and the long reverb

is a hall set to a decay time of 2.2 seconds. If the bpm of the song is still at 105, we might start with all three predelays set to 18 milliseconds and adjust them from there as the mix progresses.

Figure 8.9: A three-reverb setup
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Remember that even when using multiple reverbs, one of them (usually the longest) is usually used as a sort of glue to make all the mix elements feel as if they're being performed in the same environment in certain mix situations.

Reverb Techniques

Here are a number of techniques often used when adding reverb to particular mix elements. Don't limit yourself to just these examples, as the settings can just as easily work for other instruments in certain situations.

For Vocals:

- Automate the delay or reverb return so that in the sparse parts of the arrangement, particularly in the beginning of the song, the vocal is less wet and more upfront and intimate, which also makes the effect less obvious.

- Try mixing various reverbs. Set up three reverbs short, medium, and long (the specifics of what the actual lengths are varies with the song). On a non-ballad vocal, favor the short and medium over the long. The short (try a 0.3 to 0.6 second room or plate) one will thicken the sound. Blending in the medium (1.2 to 1.6 second plate or hall) will create a smooth transition that is quite dense but still decays fairly fast. Add a little of the longer one (2 to 3 second hall) for whatever degree of additional decay you want. The three combined will sound like one thick reverb that will stick to the vocal and not muddy it up with excess length and diffusion.

- With a singer/acoustic guitar player, try to picture the performer in an acoustic space and then realistically re-create that space around him. This lends itself to a medium-sized room or a small plate, with perhaps a little more reverb on the voice than the guitar. If the vocal is wet and the guitar dry (forgetting about leakage for a moment), it's difficult to have them both appear to share a common acoustic space.

- If a vocal effect is too prominent, bring up the reverb to where you can hear it, then back off the level 2dB. Add a dB or two at 800Hz to 1kHz to either the send or return of the reverb to bring out the effect without it being too prominent.

- For an interesting reverse reverb effect on a vocal where it's whooshing in before the vocal begins, set a reverb to a very long decay time (more than 4 seconds) and then record the reverb only onto a second track. Reverse it and move it forward on the timeline so it begins before the vocal.

For Drums:

- For the Tommy Lee "Thunder Drums" effect, set a reverb on the "cathedral" or "large hall" setting and then add a little to each drum. Pan the reverb returns so the reverb sits behind each part of the kit. For this effect to work, the bass drum has to sound tight to begin with and have a decent amount of beater present, and all the drums should be gated with the gate timed to the track. (See Chapter 9 for more on gates.)

- For an “exploding snare” type of effect, add a short slap from 50 to 125ms with a touch of feedback to the bottom snare mic. Bring the slap back on a second channel. Using an aux, send signal from both top and bottom snare mics and the slap to a short reverb of a second or less (timed to the song). By adjusting the proportions, phase, and EQ, the effect will fit it into almost any situation.

For Percussion:

- For hand percussion, such as shakers and tambourines, use a medium (0.8 to 1.2 seconds) room or plate reverb with either zero or very short (20ms) predelay.

For Guitars:

- To make guitars bigger, take a mono reverb and lower the decay time to as low as it will go (0.1 seconds if it will go that low). Pan the guitar to one side and the reverb to the other. Try different reverb types to see which works better in the song. Increase the decay time slightly to make the sound bigger or to eliminate any metallic-sounding artifacts from the reverb.

- For that early Eddie Van Halen sound, use either a chamber or a plate reverb set to about 2 seconds decay time and around 120ms predelay that’s timed to the track. Pan the guitar to one side and the reverb more to the other.

For Keyboards:

- For a keyboard pad sound that melts into the track, use a hall reverb with a 2 to 2.5 second decay and a short (20ms) predelay that’s timed to the track. Set any EQ or filters so that the extreme high and low ends are rolled off to about 8kHz and 150Hz.

For Strings:

- Use a hall reverb set to between 2.2 and 2.6 seconds with a predelay of at least 20ms timed to the track.

Using Modulation

Modulation is the third type of effect that adds dimension to a mix, although it accomplishes this more by movement than by ambience. Most musicians and engineers are very familiar with the types of modulation, but they're not clear on how they differ and when they're best used.

Types Of Modulation

There are three basic types of modulation effects: phase shift, chorus, and flange. The difference between them is that basically a chorus and flange effect comes as a result of a modulated delay that's mixed back into the original signal, with the flanger having shorter delay than a chorus. On the other hand, the phaser doesn't require a delay to achieve its effect (see Table 8.1).

All three produce a series of frequency notches that slowly sweep across the frequency bands of an instrument or vocal, leaving only a series of peaks, which is what you hear. Here's where the differences are more pronounced, as a phaser has only a small number of notches that are spaced evenly across the frequency range, while flangers and choruses have many more notches that are harmonically spaced, which provides a much more aggressive sound as a result.

Tremolo and vibrato are also popular modulation effects, although they operate differently because a delay isn't required for them to work. It's easy to get these confused, or even use their names interchangeably, but they are distinctly different; a tremolo varies the signal up and down in level, while vibrato varies the pitch up and down.

Table 8.1: The Differences Between Modulation Effects

Effect	Delay	Description
Phase shift	None	Cancels out frequencies by shifting their phase to create the effect. Frequency notches are spaced evenly across the frequency range.
Flanging	0.1ms to 5ms	The deeper the frequency cancellations, the deeper the effect. Frequency notches are randomly and harmonically spaced across the frequency response.
Chorus	5ms to 25ms	Used to thicken the sound and create a stereo image. Frequency notches are spaced harmonically across the frequency response.
Tremolo	None	Cyclicly changes the volume.
Vibrato	None	Cyclicly changes the pitch.

Flangers And Phasers

The flanger is a dramatic effect that was first created in 1966 by Ken Townsend, the chief technician at EMI Studios in London (now known as Abbey Road Studios), in an attempt to come up with something called Automatic Double Tracking, or ADT. The effect was created because

The Beatles' John Lennon loved the sound of his voice when it was doubled but hated the fact that he had to sing the song a second time, so the EMI tech staff was asked to come up with a solution. The effect was accomplished by using two tape recorders at the same time, but it led to an almost accidental discovery. By slowing one of the tape machines down by placing a finger on the tape flange (the metal part of the tape reel), a sweeping harmonics effect resulted. Lennon then coined the effect "flanging."

The first known use of flanging came on the Beatles' song "Tomorrow Never Knows," but one of the first big Top 40 hits that used the effect came a year later on a song called "Itchycoo Park" by the British group The Faces, which featured a large dose of the effect at the end of the song. Soon every artist and producer wanted flanging on their song, but there was a problem in that to get the effect, two tape recorders needed to be set up, which was both expensive and very time consuming.

Even though technology was marching along, back in the '70s the only feasible electronic simulation was an analog effect called a phaser, but the sound had none of the intensity of tape-driven real flange. That's why phasing isn't used much even today; it's just not that dramatic of an effect.

It wasn't until digital delays came on the market in the '80s that it became possible to simulate true tape flanging, and now just about every modulation plugin and stomp box can do at least a passable simulation of the effect if set up correctly.

Chorus

What makes a chorus different from a flanger is that the delay is longer, going from about 5 to 25ms, and the frequency notches needed to create the effect occur in a fixed cycle. The effect shines in stereo and can really widen the sound of a track quite a bit. Since chorusing first was introduced by Roland in 1980, many hits from that era used the effect over and over.

Today you'll find that most modulation plugins and hardware allow you to easily select between chorus, phasing, and flanging, since they're all related, but the ones that are used in stereo are the most dramatic.

Tremolo And Vibrato

For years, guitar amps included tremolo as standard feature, although in some cases (such as on Fender amps) it was mislabeled as vibrato. As stated previously, they're not the same, since tremolo is a cyclic variation in volume, while vibrato changes the pitch of the sound.

Guitars weren't the only instrument to use tremolo, as both the original Rhodes and Wurlitzer electronic pianos had the effect built in. Vibrato is rarely used because the variation in pitch can make the track or other tracks seem out of tune.

Typical Modulation Setups

While flanging, tremolo, and vibrato are pretty dramatic effects and can cause a mix element to stand out, chorusing is often used to widen a track. That said, modulation effects aren't usually set up in advance on a dedicated set of sends and returns, since the effect tends to work better on an individual instrument or vocal.

That said, sometimes a flanger is required across the entire mix, so it's best inserted either across the stereo buss or across a separate subgroup so the level can be easily automated and adjusted (see Figure 8.10).

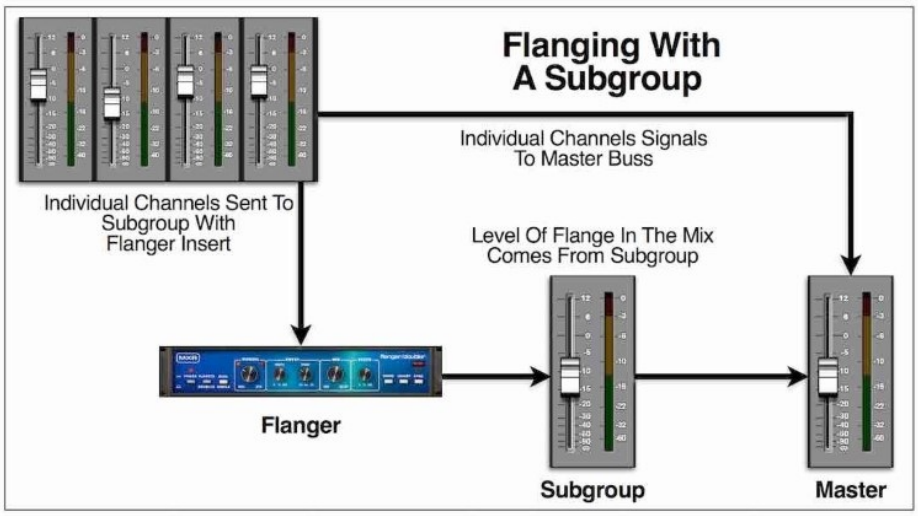

Figure 8.10: A flanger inserted on a subgroup
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Modulation Techniques

Here are a few techniques often used when adding modulation to a mix element. Don't be afraid to experiment, though, because any of these techniques may work well in other mix situations as well.

For fatter lead or background vocals:

- Use some chorusing panned hard left and right to fatten up the sound. Ride the chorusing effect, adding and subtracting it according to what sounds best.

For out-of-tune vocals:

- If you have something against Auto-Tune or just want to cover up an out-of-tune vocal, use a stereo chorus or flanger and pan it slightly left and right. The more out of tune the vocal, the more modulation you need to cover it up. This does an effective job of taking the listener's attention off any sour notes.

- Pan a delayed signal behind the vocal and then send it to a chorus, detuning both sides a bit so the delay sounds wide and the modulation steals your attention from the tuning.

Robot voice:

- Use a deep flange to make the voice sound metallic and then use it as an external key to gate SMPTE code in time with the voice. Mix in the gated code gently.

EQing Effects

One of the things that many mixers struggle with is getting reverb and delay effects to blend well in the mix. This happens more with reverbs than delays, especially during those times when the reverb just never seems to sound quite right. Usually the way the problem is addressed is to audition different reverb presets until something is found that seems to work better, but that can take time, and it's easy to end up chasing your tail to the point where you're never sure which preset actually sounds the best. What many seem to forget is that most of those presets are the same basic reverb with different EQ settings, which you can add yourself to get there faster.

One thing that happened regularly back in the early days of analog reverb (especially with plates) is that one way to tune a reverb to the track was to insert an EQ on the send before the actual reverb itself. Usually the EQ was set more to cut than to boost (although you'd boost it if you wanted a bright-sounding plate that jumped out of the mix), but if done well, the reverb would suddenly fit a lot better in the track. In fact, back in the classic days of the big studios, this was done in the back room and not left up to the engineer at the console, and it became one of the reasons for clients wanting to work there; they loved the sound of their reverbs.

We can use those same techniques today using reverb plugins on our DAW, or any other effects for that matter. Just remember that it usually sounds best if the EQ is placed before the reverb, not after it, because it has a great effect on the frequency response of the reverb.

> ***TIP**: Try adding tape-saturation plugins such as Avid's Heat or Universal Audio's Studer A800 Tape Recorder to the effect send or return. The extra harmonics sometimes give it more depth.*

The following sections describe three EQ curves that are frequently used.

On Vocals

One thing about reverb is that any low end from it just muddies up the track, and any high end may stick out too much, which is why it might be a good idea to roll each end of the frequency spectrum off a bit. In many cases this means somewhere around 200Hz and 10kHz (or even lower). When reverb is used on vocals, sometimes it fits better if there's also a bit of an EQ scoop in the midrange around 2kHz, where the consonants of the vocal live, so the effect stays out of the way frequency-wise (see Figure 8.11). Once again, this is very effective on delays and modulators as well.

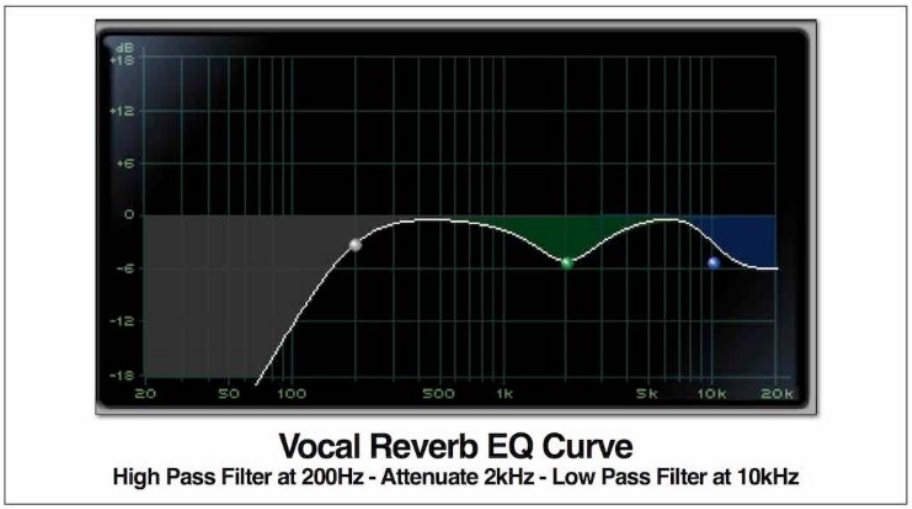

Figure 8.11: Effects EQ for vocals
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On Instruments

For instruments, the Abbey Road curve, which is what the famous studio has used on their reverbs since the ’60s, works very well. This means that the low end is rolled off at 600Hz and the high end at 10kHz (see Figure 8.12). This curve makes any reverb sound a lot smoother and fit better with the track. You’ll find that this setting just increases the depth without it sounding washed out when you add more reverb using this curve. Of course, too much of a good thing is no good either, so be judicious with the amount you add.

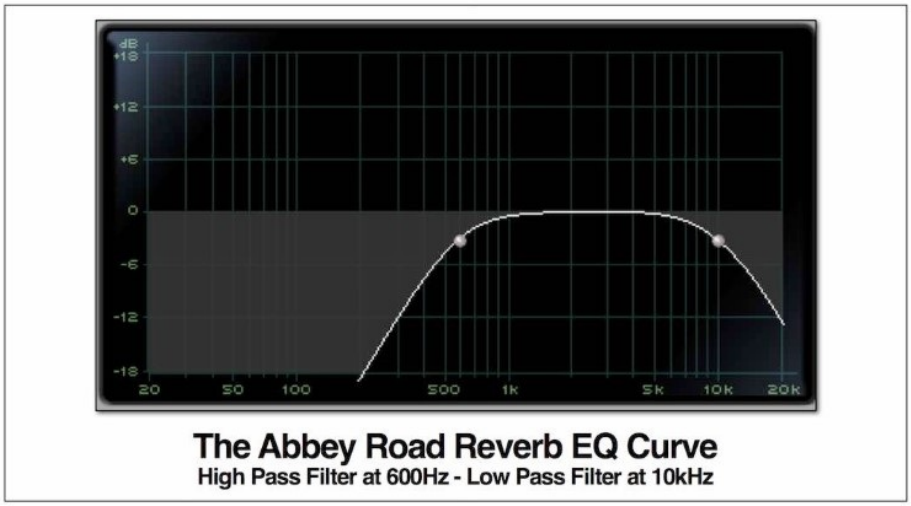

Figure 8.12: The “Abbey Road” effects EQ for instruments
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On Drums

Sometimes reverb on the drums is the toughest of all in that you want depth without calling attention to the ambience. A good way to do that is a variation of the Abbey Road curve where the high end is severely rolled off to 6k, 4k, or even 2kHz (see Figure 8.13)! You'll find that you'll have some depth without the ambience ever calling attention to itself.

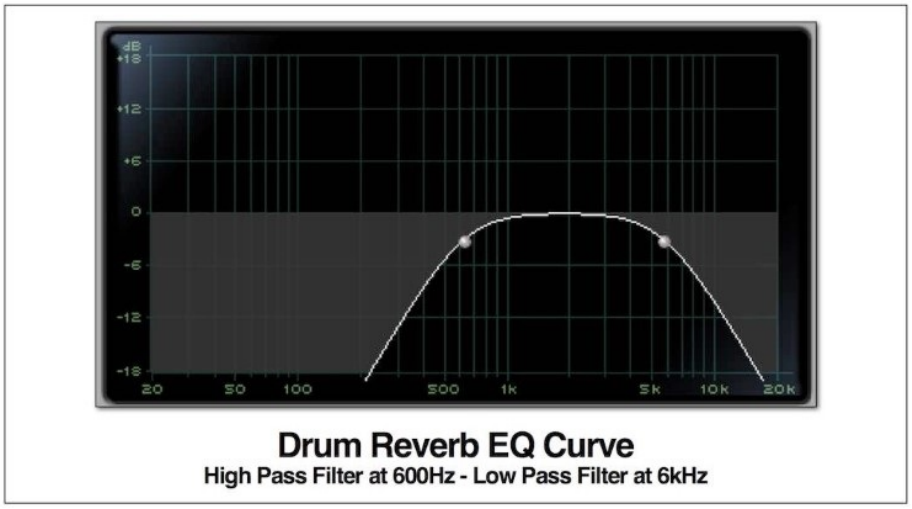

Figure 8.13: Effects EQ for drums
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While these EQ curves work great with reverbs, don't be afraid to try them with delays or modulation effects, as the results are very similar. You'll get depth without the delay or effect getting in the way. Of course, if you want to really hear the reverb or delay, go the opposite way and increase the high end, and the effect will jump right out of the track.

Equalization Tips For Reverbs And Delays

- To make an effect stick out, brighten it up.
- To make an effect blend in, darken it up (filter out the highs).
- If the part is busy (such as with drums), roll off the low end to keep it out of the way.
- If the part is open, add low end to the effect to fill in the space.
- If the source part is mono and panned hard to one side, make one side of the stereo effect brighter and the other darker and pan the brighter side opposite the source track.

Layering Effects

Layering means that each instrument or element sits in its own ambient environment, and each environment is usually artificially created by effects. The idea here is that these sonic atmospheres don’t clash with one another, just like in the case of frequency ranges.

The following section features some suggestions so that the sonic environments don’t clash.

Layering Tips For Reverbs And Delays

- Layer reverbs by frequency, with the longest being the brightest and the shortest being the darkest. Pan the reverbs any way other than hard left or right.
- Return the reverb in mono and pan accordingly. All reverbs needn’t be returned in stereo.
- Get the bigness from reverbs and depth from delays, or vice versa.
- Use a bit of the longest reverb on all major elements of the track to tie all the environments together.

“My personal taste is to use more layers, like using several reverbs to create one reverb sound, or using several short and long delays. My reverbs and effects usually end up coming from four to eight different sources. They’ll be short, long, bright, dull, and everything you need to make an environment.”

—Bob Bullock

Reamping

One of the ways to re-create a natural environment is by a process known as reamping. This is accomplished by actually sending a signal of an already-recorded track (say a guitar or keyboard) back out to an amplifier in the studio, then miking it from a distance to capture the ambience of the room (see Figure 8.13). It's all the better if the ambience is captured in stereo.

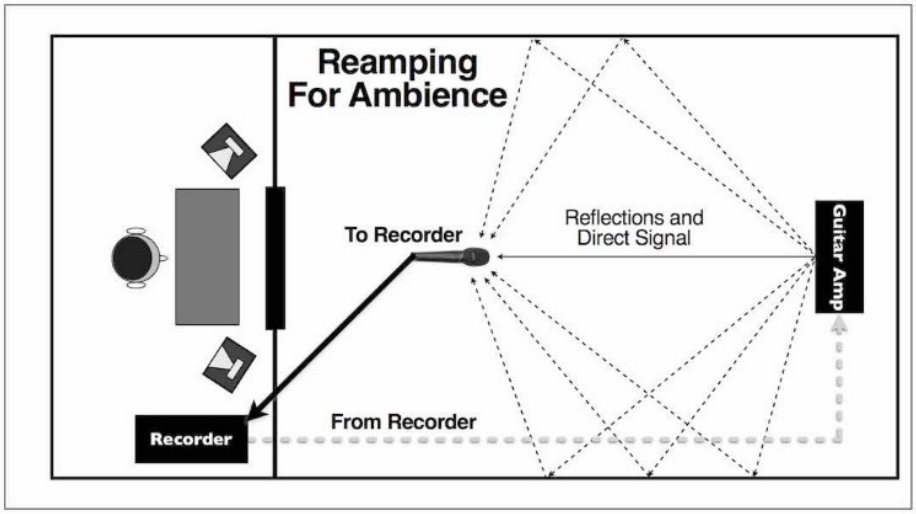

Figure 8.14: Reamping to achieve natural ambience
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> *"What I will do frequently when we're layering with synths is to add some acoustics to the synth sounds. I think this helps in the layering in that the virtual direct sound of most synthesizers is not too interesting, so I'll send the sound out to the studio and use a coincident pair of mics [X/Y stereo pair] to blend a little bit of acoustics back with the direct sound. It adds early reflections to the sound, which many reverb devices can't do."*
>
> *—Bruce Swedien*

Chapter 9

The Dynamics Element: *Compression, Limiting, Gating, And De-Essing*

At one point in recording history, the control of the volume envelope of a sound (or its dynamics) would not have been included as a necessary element of a great mix. In fact, dynamics control is still not a major component in some of the orchestral and jazz mixing world. Today's modern mixes have different demands, though, so the manipulation of dynamics plays a major role in the sound in most music, because the fact is that almost nothing else can affect your mix as much and in so many ways as compression.

> *"I think that the sound of modern records today is compression. Audio purists talk about how crunchy compression and EQ are, but if you listen to one of those jazz or blues records that are done by the audiophile labels, there's no way they could ever compete on modern radio even though they sound amazing. Every time I try to be a purist and go, 'You know, I'm not gonna compress that,' the band comes in and goes, 'Why isn't that compressed?'"*
>
> *—Jerry Finn*

Types Of Dynamics Control

An audio source’s dynamic range is controlled by the use of compression, limiting, downward expansion (gating) and upward expansion (transient design). For those of you new to mixing or for those who need a review or clarification, here’s a brief description of each. See the glossary or any number of recording texts for more complete information.

Compression

What we know as compression is more properly called dynamic range compression because it's the process of taking an audio source signal with a large dynamic range and making it smaller. This is done by lowering the loudest portions of the program and increasing the lowest ones so the volume level is more constant.

Compressors work on the principle of gain ratio, which is measured on the basis of input level to output level, and is set by using the Ratio control (see Figure 9.1). As an example, a ratio of 4 to 1 or 4:1 means that for every 4dB that goes into the compressor, 1dB will come out (see Figure 9.2). If the ratio is set at 8:1, then for every 8dB that goes into the unit, only 1dB will be seen at the output. A ratio of 1:1 results in no compression at all.

Figure 9.1: Typical compressor parameters
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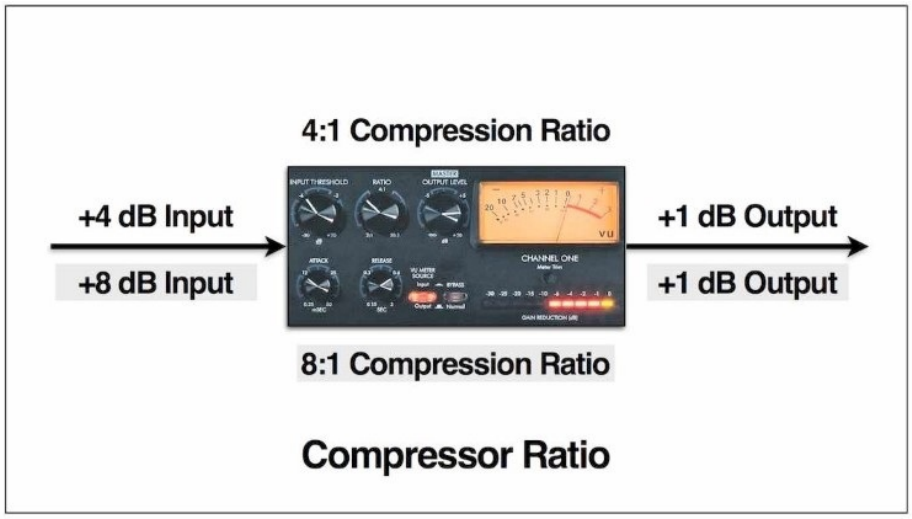

Figure 9.2: Compressor ratio
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The Threshold control determines the point in the signal level where the unit will begin to

compress (see Figure 9.1). As a result, threshold and ratio are interrelated, and one will affect the way the other works. Some compressors (such as the Universal Audio LA-3A in Figure 9.3) have a fixed ratio, but on most units the parameter is variable.

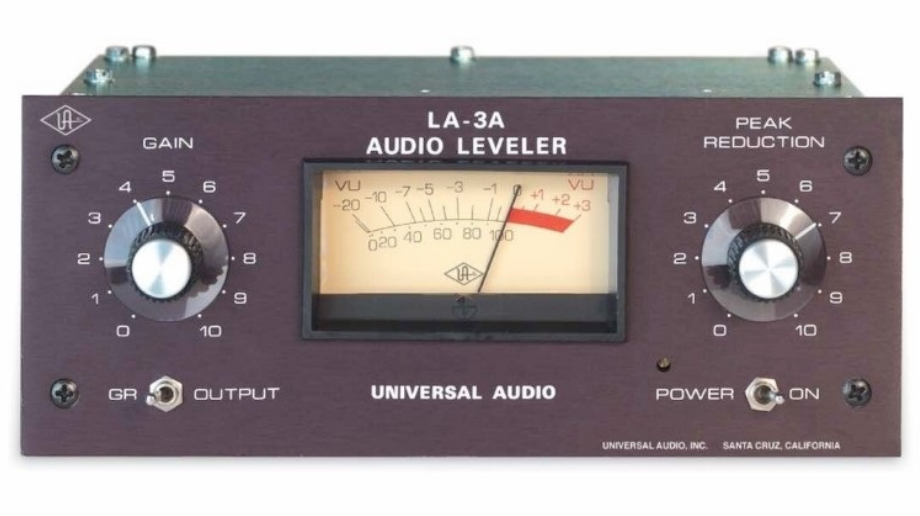

Figure 9.3: A Universal Audio LA-3A
Courtesy Universal Audio

Most compressors have Attack and Release parameter controls (see Figure 9.1) that determine how fast or slow the compressor reacts to the beginning (the attack) and end (the release) of the signal envelope. These controls are especially important because the setting is crucial to how the compression works and sounds. Because the settings on the Attack and Release controls can be tricky if you're not sure how to use them, some compressors have an Auto mode, which will set the attack and release according to the dynamics of the input signal. Although Auto mode can work relatively well, it doesn't allow for the precise settings that may be required to properly control certain source material. While most compressors have some control over the volume envelope, some compressors have a fixed attack and release (such as the dbx 160 series—see Figure 9.4), which gives it a particular sound as a result.

Figure 9.4: A dbx 160x
Courtesy Harmon International Industries

Many compressors also have a Knee parameter, which sets how fast the compressor will begin to compress after the signal reaches the threshold level (see Figure 9.1). A low value (sometimes it's measured in dB, so 0dB would be the lowest) means that the compression will begin instantly at threshold. A higher setting will gradually ease in the compression, which may sound better on some types of program material where the compression is meant to be less apparent.

When a compressor operates, it actually decreases the gain of the signal, so there's another control that allows the signal to be boosted back up to its original level or beyond called Make-Up Gain or Output (see Figure 9.1).

All compressors have a gain-reduction meter to show how much compression is occurring at any given moment (see Figure 9.1). This can look like a normal VU or peak meter, but it reads backwards. In other words, 0dB means that the signal is below threshold and no compression is taking place, and any travel to the left or down into the minus range shows the amount of compression that's occurring. For example, a meter that reads -6dB indicates that there is 6dB of compression taking place at that time.

Many compressors also have a feature known as a Sidechain (sometimes called a "key" input), which is a separate input into the compressor so that other signal processors can be connected to it (see Figure 9.1). This sidechain can have a number of useful purposes, such as when an EQ is connected to it to make a makeshift de-esser. If only the frequencies in the upper range are boosted, the loud "S" sounds from a vocalist will be attenuated when they exceed the compressor's threshold (more on that in the "De-essing" section of this chapter). You can also connect a delay, reverb, or any other signal processor to the sidechain to create some unusual, program level–dependent effects. Because a sidechain isn't needed for most everyday compressor operations, many manufacturers elect not to include sidechain connections on hardware units, but may include one on the plugin version.

A sidechain can also be used to "duck" or attenuate another instrument when a new one enters. For instance, if you connected a send of a vocal track into the sidechain of a compressor inserted on a loop track, the loop would lower in volume whenever the vocal entered and then return to its normal level when the vocal stopped (see Figure 9.5). This is what happens at an airport when the music is automatically lowered for a gate announcement and then returns to its normal level when the announcement is finished.

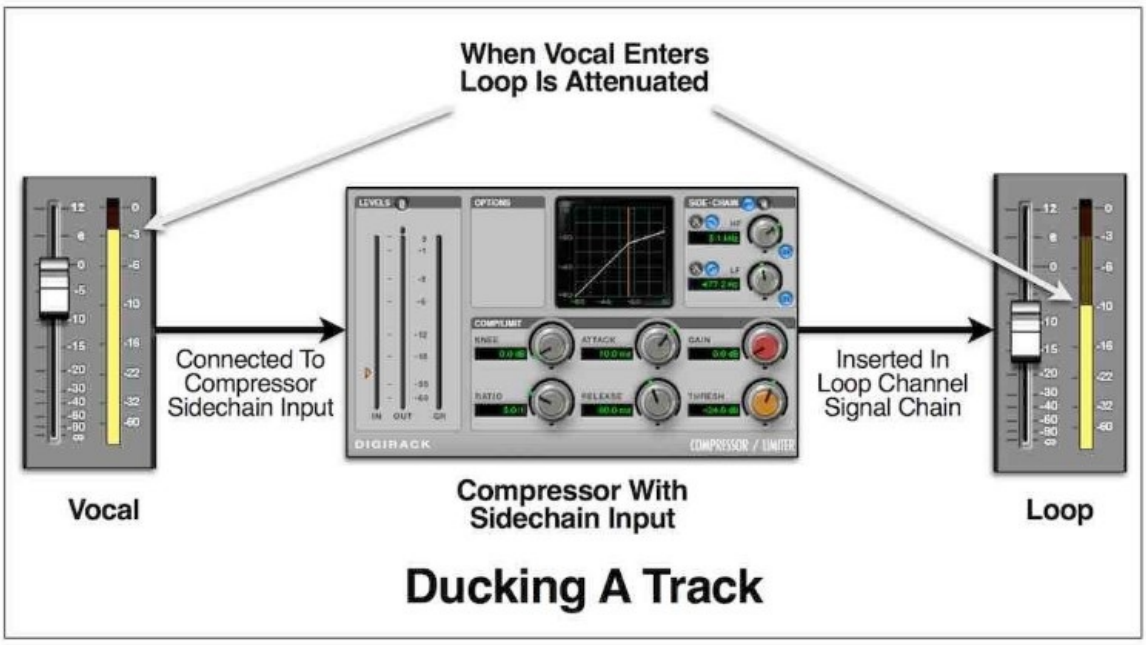

Figure 9.5: Connecting to the compressor sidechain to duck a track
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Compressor Differences

In the days of analog hardware compressors, there were four different electronic building blocks that could be used to build a compressor. These were:

- **Optical**. A light bulb and a photocell were used as the main components of the compression circuit. The time lag between the bulb and the photocell gave it a distinctive attack and release time (like in an LA-2A).
- **FET**. A Field Effect Transistor was used to vary the gain, which had a much quicker response than the optical circuit. (A Universal Audio 1176 is a good example.)
- **VCA**. A Voltage Controlled Amplifier circuit was a product of the ‘80s and had both excellent response time and much more control over the various compression parameters (The dbx 160 series is an example of a VCA-type compressor, although some models didn’t have a lot of parameter controls).
- **Vari-Gain.** The Vari-Gain compressors are sort of a catch-all category because there are other ways to achieve compression besides the first three (such as the Fairchild 670 and Manley Vari-Mu).

As you would expect, each of the above has a different sound and different compression characteristics, which is the reason why the settings that worked well on one compressor type won’t necessarily translate to another. The good thing about living in a digital world is that all of these different compressor types have been duplicated by software plugins, so it’s a lot easier (not to mention cheaper) to make an instant comparison on a track and decide which works better in a particular situation.

Multi-Band Compression

Multi-band compression splits the input audio signal into two or three frequency bands, each with its own compressor. The main advantage of a multi-band is that a loud event in one frequency band won’t affect the gain reduction in the other bands. That means something like a loud kick drum will cause the low frequencies to be compressed, but the mid and high frequencies are not affected. This allows you to get a more controlled, hotter signal with far less compression than with a typical single-band compressor.

Limiting

Compression and limiting are closely related, the main difference being the setting of the ratio parameter and the application. Any time the compression ratio is set to 10:1 or greater, the result is considered limiting, although most true limiters have a very fast attack time as well. A limiter is essentially a brick wall level-wise, allowing the signal to
get only to a certain point and little more beyond. Think of it doing the same thing as a speed governor that's sometimes used on 18-wheel trucks owned by a trucking company to make sure that they're not driven beyond the speed limit. Once you hit 65 mph (or whatever the speed limit in your state is), no matter how much more you depress the gas pedal, the truck won't go any faster. The same theoretically occurs with a limiter. Once you hit the predetermined level, no matter how much you try to go beyond it, the level pretty much stays the same.

Most modern digital limiters (either hardware or software) have an internal function known as "look ahead" that allows the signal detection circuitry to look at the signal a millisecond or two before it hits the limiter (Figure 9.6). This means that the limiter acts extremely fast and just about eliminates any overshoot of the predetermined level, which can be a problem with analog limiters because they react much more slowly to transients.

Figure 9.6: Universal Audio Precision Limiter
Courtesy Universal Audio

Limiting is used a lot in sound reinforcement for speaker protection (there are some limiters on powered studio monitors as well), and not as much in mixing with the following exception. Many engineers who feel that the bass guitar is the anchor for the song want the bass to have as little dynamic range as possible. In this case, limiting the bass by 3 to 6dB (depending on the song) with a ratio of 10:1, 20:1, or even higher will achieve that.

De-Essing

One of the major problems when tracking vocals is a predominance of S's that comes from a combination of mic technique, the mic used (usually a condenser with a presence peak in the sibilance range), the EQ added to make it cut through the mix, and the use of heavy compression. Sometimes this might not be too much of an issue until it's time to mix, but when a compressor is put on the vocal to even out the level, all of a sudden every "S" from the singer seems ear-piercing. This effect is what's known as sibilance and is totally undesirable.

The way to combat sibilance is to use a de-esser, which is a unit that compresses just the S frequencies that usually fall somewhere between 3kHz and 10kHz, depending upon the situation.

A de-esser can be created from a compressor with an equalizer plugged into the sidechain as stated above, or it can be a dedicated unit designed just for this purpose. While many hardware de-essers are limited to two parameter controls, Threshold and Frequency (Figure 9.7), software de-essers are much more sophisticated, allowing for pinpoint control that a hardware unit just can't duplicate. The Threshold control is similar to the control found on a compressor/limiter in that it sets the level when the de-essing process begins. The Frequency control allows you to fine tune the frequency to exactly where the S's occur.

Figure 9.7: A Pro Tools Digirack de-esser
Courtesy Avid

Gating

Although not used nearly as much in the studio now that so much processing and editing is possible in digital workstations, gates are still used in certain situations in the studio and a lot in sound reinforcement (see Figure 9.8). More correctly called a downward expander, a gate keeps a signal turned off or at a lower level until it reaches a predetermined threshold level, at which time it opens and lets the sound through. The gate can be set to mute the sound completely when it drops below threshold or to just lower the level to a predetermined amount. Depending on the situation, just turning the level down a bit many times sounds more natural than turning it completely off, although the total silence can sometimes be used as a great effect.

Figure 9.8: A Pro Tools Digirack gate
Courtesy Avid

For a long time engineers would use a gate (sometimes called a noise gate or expander) to reduce or eliminate problems on a track such as noises, buzzes, coughs, or other low-level noises off-mic. On loud electric guitar tracks, for instance, a gate could be used to effectively eliminate amplifier noise during the times when the guitar player is not playing. Because most of these situations can now more effectively be dealt with in the DAW (see Chapter 11, “Advanced Techniques,” for how it’s done), gates aren’t used as much in the studio, although there’s still a need for them in a live-performance situation.

Gates are still frequently used on drums to control leakage resulting from tom mics in a mix, or to tighten up the sound of a floppy kick drum by decreasing the ring after it’s struck by the beater.

Gates can also have a sidechain, or just an additional input called a key or trigger input, which allows the gate to open when triggered from another instrument, channel, or processor. This can be very useful in a number of situations, as seen in the “Gating Techniques” section at the end of this chapter.

Just like compressors, modern hardware gates are very fast, but plugins have the advantage that they can be designed so that the sidechain senses the signal a millisecond or two before it arrives at the gate's main input, allowing it to begin opening just before the transient arrives. This look-ahead facility is only an advantage when dealing with sounds that have a very fast attack, so it's sometimes switchable or variable when it is provided.

Transient Shapers

The opposite of a downward expander is an upward expander, sometimes known as a transient designer or shaper. An upward expander takes signal above the threshold and increases its gain, acting like the opposite of compression.

With a normal compressor with 2:1 ratio, for every 2dB above threshold the output would be 1dB. With an upward expander, that ratio is reversed to 1:2, so for every 1dB above threshold the output will rise by 2dB. This increases the dynamic range by making loud parts or peaks louder, and has the effect of putting the transients back into a part that's been too compressed. This is often used on drum overheads, room mics and snare drums to emphasis the crack of the drum and lengthen its effect by increasing the Release control.

Some excellent transient shaper plugins include the Stillwell Audio Transient Monster, and the SPL Transient Designer.

Using Compression

If there is one major difference between the sound of a demo or semi-pro recording and a finished professional mix, it's the use of compression. As a matter of fact, the difference between the sound of one engineer's mix from another is more often than not how he uses his compressors.

There are two reasons to add compression to a track or mix: to control the dynamics or as an effect.

Controlling Dynamics

Controlling dynamics means keeping the level of the sound even. In other words, lifting the level of the soft passages and lowering the level of the loud ones so that there's not too much of a difference between them.

Here are a couple of instances where this might be useful:

- **On a bass guitar.** Most basses inherently have certain notes that are louder than others and some that are softer than others depending upon where on the neck they're played. Compression can even out these differences.
- **On a lead vocal**. Most singers can't sing every word or line at the same level, so some words may get buried as a result. Compression can help every word to be heard.
- **On a kick or snare drum.** Sometimes the drummer doesn't hit every beat with the same intensity. Compression can make all hits sound somewhat the same.

***TIP**: When controlling dynamics, usually a very small amount of compression (2 to 4dB) is used to limit the peaks of the signal.*

> *"I like to compress everything just to keep it smooth and controlled, not to get rid of the dynamics. Usually I use around a 4:1 ratio on pretty much everything I do. Sometimes on guitars I go to 8:1. The kick and the snare I try not to hit too hard because the snare really darkens up. It's more for control, to keep it consistent. On the bass, I hit that a little harder, just to push it up front a little more. Everything else is for control more than sticking it right up in your face."*
>
> *—Benny Faccone*

Compression As An Effect

Compression can also radically change the sound of a track. A track compressed with the right compressor and with the correct settings can make it seem closer to the listener and have more aggression and excitement. The volume envelope of a sound can be modified to have more or less attack, which can make it sound punchy, or to have a longer decay so it sounds fatter.

> *"Compression is the only way that you can truly modify a sound because whatever the most predominant frequency is, the more predominant that frequency will be the more you compress it. Suppose the predominant frequencies are 1k to 3k. Put a compressor on it, and the bottom end goes away, the top end disappears, and you're left with "Ehhhhh" [a nasal sound]."*
>
> *—Andy Johns*

> *"I usually spend the last two hours of my mix not doing much mixing but listening and then making little adjustments to the master buss compressor and hearing what the impact is to all the parts. Most of the effect is how it's putting the low- frequency information in check, which it does without the meter even moving at all. Even when you're hitting it very lightly, it still has a dramatic effect on the music."*
>
> *—Bob Brockman*

Placement In The Signal Chain

Where a compressor, limiter, de-esser, gate or expander is placed in the signal chain has a dramatic effect on what that device does to the sound. If a compressor or limiter is placed after an EQ, every time the EQ is changed it will affect the compressor settings. Plus, any frequency that's boosted will have less of an effect because it will be turned down by the compressor. Likewise, a de-esser placed after an EQ may become ineffective if the midrange or high frequencies are changed. Similarly, the threshold of a gate can change if the EQ is changed if it's placed before it in the signal chain.

That's why it's best to place any compressor, limiter, or de-esser, first in the signal chain, because the signal will receive all the benefits of the dynamics module and any other processor that comes after it (see Figure 9.9).

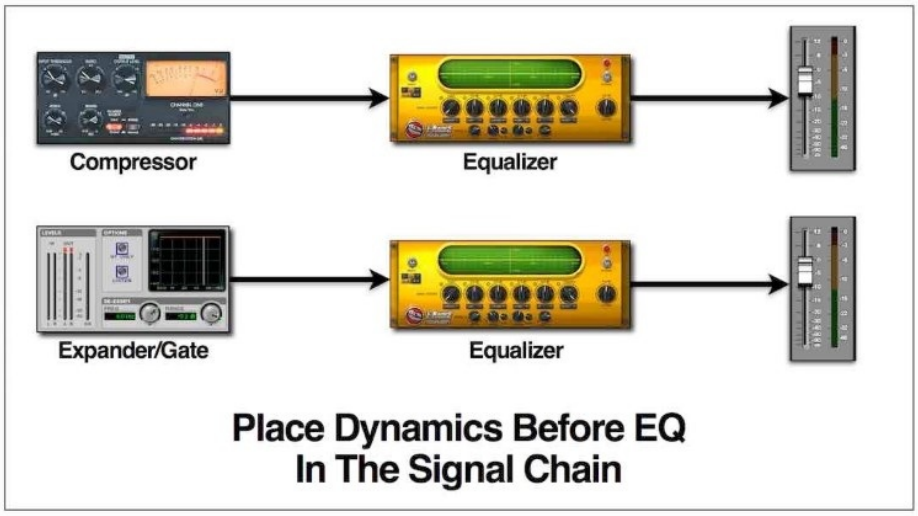

Figure 9.9: Dynamics placement in the signal chain
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Setting The Compressor

In most modern music, compressors are used to make the sound "punchy" and in your face. The trick to getting the punch out of a compressor is to let the attacks through and play with the release to elongate the sound. Fast attack times can sometimes reduce the transients, and therefore the high end of a signal, while slow release times may make the compressor pump out of time with the music or even cause the signal to distort.

Since the timing of the attack and release is so important, here are a few steps to help set up a compressor. Assuming you have some kind of constant tempo to the song, you can use the snare drum to set up the attack and release parameters, which may also transfer to other tracks as well. Regardless of the instrument, vocal, or audio source, the setup is basically the same.

1. Start with the attack time set as slow as possible (probably around 500ms), the release time set as fast as possible (probably around 0.1 or 0.2 seconds), and the threshold set as high as possible so no compression is triggered.
2. Decrease the threshold until the meter shows some compression, then turn the attack faster until the sound of the instrument begins to dull, which now means that you're compressing the transient portion of the sound envelope. Stop increasing the attack time at this point and even back it off a little so the sound stays crisp and maintains its clarity.
3. Adjust the release time so that after the initial attack, the volume goes back to at least 90 percent of the normal level by the next beat. If in doubt, it's better to have a shorter release than a longer one, although you may hear the compressor pumping if it's set too fast.
4. If you want the compression to sound smooth and controlled, select a low ratio of around 2:1. If you want to hear the compressor or you want the sound to be punchier, select a compression ratio above 4:1.
5. Bypass the compressor to see whether there's a level difference. If there is, increase the Gain or Output control until the volume is the same as when it's bypassed.
6. Add the track to the rest of the mix and listen. Make any slight adjustments to the attack and release times as needed.

***TIP**: The idea of setting the compressor's attack and release is to make it breathe with the pulse of the song.*

> *"I get the bass and drums so they just start to pump to where you can actually hear them breathing in and out with the tempo of the song. What I'll do is put the drums and bass in a limiter and just crush the hell out of it. Then I'll play with the release and the attack times until I can*

actually make that limiter pump in time with the music, so when the drummer hits the snare, it sucks down and you get a good crest on it. When he lets go of the snare, the ambience of the bass and the drums suck and shoot back up again. You can actually hear a [breathing sound] going on that was never there before. It really was there; it's just that you're augmenting it by using that limiter."

—Lee DeCarlo

"I set the attack as slow as possible and the release as fast as possible so all the transients are getting through and the initial punch is still there, but it releases instantly when the signal drops below threshold. I think that's a lot of the sound of my mixes. It keeps things kinda popping the whole time. Also, you can compress things a little bit more and not have it be as audible."

—Jerry Finn

What's The Right Amount Of Compression?

The amount of compression added is usually to taste, but generally speaking, the more compression the greater the effect will be and the more likely it will change the sound of the instrument or vocal. Compression of 6dB or less is meant more for controlling the dynamics of a track rather than for changing its sonic quality, and often just a dB or two is all that's needed.

That said, it's not uncommon for radical amounts of compression to be used sometimes. Ten or even 20dB is routinely used for electric guitars, room mics, drums and even vocals. As with most everything else, it depends on the song, the arrangement, the player, the room, and the gear.

> *"There are times when there's singing when they're not in compression at all, but if my limiter hits 15 or 20dB of compression and I don't hear it, I don't think about it for an instant more."*
>
> *—Nathaniel Kunkel*

***TIP**: The higher the compressor ratio control is set, the more likely you'll hear the compressor or limiter work. The more compression that's added, the more likely it is that you'll hear it.*

Parallel Compression

Parallel compression is a trick that mixers have been using ever since the ‘70s to make a track sound punchy in a very natural way. The trick involves sending the signal to another channel either through a mult on a hardware console or via a buss or aux on a DAW mixer, then adding a compressor to the second channel only (see Figure 9.10).

The compressed channel is then brought up in level so it’s just under the original non-compressed channel. This gives the track a feeling of control without sounding too squashed. Keep in mind that parallel compression works well with just about any individual instrument or vocal, and if you’re working in a DAW, you’re limited only by the number of tracks that you want to deal with.

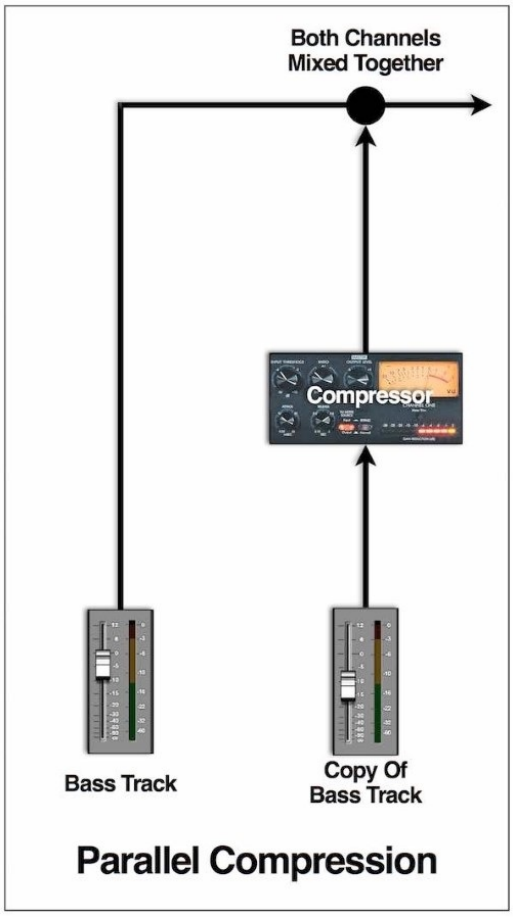

Figure 9.10: Parallel compression
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> *“On this mix right now there’s a parallel compressor on the kick and snare, then there’s another just on the snare. There’s a stereo one on the toms and overheads, a mono one on just the dirty bass (this song has three basses), a stereo one on the guitar and vocals, and then a couple of different ones just for the lead vocal, one that’s sort of spitty and grainy and one that’s sort of fat. That changes from mix to mix.”*

—Andrew Scheps

The New York Compression Trick

One of the little tricks that seem to set New York mixers apart from everyone else is something I call the "New York Compression Trick." It seems as if every mixer who's ever mixed in New York City comes away with this parallel compression maneuver. Back in the early days of recording when there were on a few compressors in each studio, this was a way to compress the entire rhythm section with just a single unit. Even if you don't mix in NYC, once you try it you just might find yourself using this trick all the time, because it's indeed a useful method to make a rhythm section rock.

Here's the trick:

1. Route the drums, and even the bass, via a send or buss to a stereo compressor or two DAW channels with the compressor inserted.
2. Adjust the threshold of the compressor so there's at least 10dB of compression. (Use even more if it sounds good.)
3. Return the output of the compressor to a new pair of input channels. (No need to do this if you're using a DAW since you already have them set up.)
4. Add some high end (6 to 10dB at 10kHz) and low end (6 to 10dB at 100Hz) to the compressed signal. (This is optional, as you'll still get the sound without the EQ.)
5. Bring the fader levels of the compressor channels up until they're tucked just under the present rhythm section mix to where you can just hear it.

The rhythm section will now sound bigger and more controlled without sounding overly compressed.

One of the cooler things about many of the new generation of compressor plugins is that they have "blend" controls which lets you set up parallel compression without having to go through some of the gyrations of above. Although you still have to use much of the same routing for the New York Compression trick, simple track parallel compression is now a snap.

> *"What I will do a lot is buss my drums to another stereo compressor and blend that in just under the uncompressed signal. Sometimes if everything sounds good but the bass and kick drum aren't locked together or big enough to glue the record together, I'll take the kick and bass and buss them together to a separate compressor, squish that a fair amount, and blend it back in. I'll add a little bottom end to that if the record still isn't big enough on the bottom. This helps fit the bass and kick lower on the record and gets it out of the way of the vocal."*
>
> *—Joe Chiccarelli*

Compression On Individual Instruments

Back in the early days of recording, a studio was lucky to have more than a couple of compressors available, which meant that you had to be judicious in what you used them on. This all changed as recording became more and more sophisticated, even to the point where eventually large-format consoles came with dynamics on every channel.

Now when it comes to adding dynamics in a DAW, the limiting factor is the processing power available in the computer. While it may not be necessary or even desirable to add compression on every track in a session, sometimes tracks can benefit from just a slight touch. Let's look at the effect compression can have on individual music tracks.

> *"To me, the key to compression is that it makes the instrument sound like it's being turned up, not being turned down. If you choose the wrong compressor or you use it the wrong way, then your stuff can sound like it's always going away from you. If you use the correct compressor for the job, you can make it sound like, 'Man, these guys are coming at you.' It's very active and aggressive."*
>
> *—Ed Seay*

A Drum Compression Primer

One thing that most modern mixes call for is a punchy drum sound, and this can be achieved with compression, although a great acoustic drum sound, a great drum, and a great recording are certainly all major components to the sound. It would be great if every drummer hit every beat on the kick and snare with somewhat the same intensity, but unfortunately that doesn't even always happen with the best session drummers you can find.

Some intensity changes are both proper and natural in music, but when the intensity noticeably changes from beat to beat, the pulse of the song can feel erratic and sometimes even a slight change in level can make the drums feel a lot less solid than they should be. Compression works wonders to even out those erratic hits and helps to push the kick and snare forward in the track to make them feel punchier. Let's take a look at how to do that with the drums.

Compressing The Kick And Snare

The biggest question that arises when compressing either the kick or the snare is, "How much is enough?" This depends first and foremost on the sound of the drum itself and the skill of the drummer. A well-tuned drum kit that sounds great in the room should record well, and a reasonably good drummer with some studio experience usually means that less compression is needed because the hits are fairly even to begin with.

Even a great drummer with a great sounding kit can sometimes benefit from a bit of compression, though, and as little as a dB or two can work wonders for the sound in many situations. With only that amount, the setup of the compressor is a lot less crucial, especially the attack and release.

> ***TIP**: A ratio setting of 2:1 sometimes provides the most punch as it acts on the transients less.*

Sometimes you need the kick or snare to cut through the mix and seem as if it's in your face, and that's when 3 to 6dB or so does the job. It's here that the setup of the compressor is critical because you're imparting its sound on the drum. Make sure to tweak the Attack and Release controls, as outlined earlier, as well as the Ratio control, and even try a number of different compressors. You'll find they all react differently, even with the same settings, so it's worth the time to experiment.

> ***TIP**: If the attack is set too fast, the drum will sound less punchy, regardless of how much or how little compression you use.*

Compressing The Room Mics

Room ambient mics are meant to add the "glue" to the sound of a kit and can really benefit from a fair amount of compression, which means anywhere from 6 to 10dB. In fact, many mixers prefer

the room sound to be extremely compressed, with way more than 10dB being the norm.

The problem is that the more compression you use, the more the ambience of the room is emphasized. That's okay if the room tracks were recorded in a great sounding acoustic space, but if it has a lot of bad sounding reflections and the ceiling is low, you may be emphasizing something that just doesn't add much to the track.

> ***TIP**: If the ambience on the room mic tracks sounds bad, set the attack time so it's much shorter than usual to cut off the sound of the initial drum transient, and keep the release time short so the ambience isn't emphasized, then tuck the room tracks in just under the other drum tracks.*

Note that regardless of how good the room mics sound, the more of them you use, the less space there will be for the other instruments in the track. The more instruments there are, the more you'll have to back them off. Sad but true, but unfortunately, there's only so much sonic space to any mix.

Compressing Vocals

Most singers aren't able to sing every word or line at the same level, and some words get buried in the mix as a result. Compression helps to even out the level differences so it's easier to hear every word.

Depending on the vocal style, the song and arrangement, the mic technique, and any number of other factors, the amount of compression required on a vocal can vary a great deal. Some vocalists are so consistent that only a dB or two is needed, but it's not uncommon to use as much as 10dB or more on some vocals either. Here's how to set up the compressor on a vocal.

1. Solo the vocal and insert a compressor on its channel. Begin with the ratio set to 4:1, but experiment with higher ratios if you're not getting the sound or control that you're looking for. You can also use a 2:1 ratio with more compression to make it sound smoother with less of the sound of the compressor.
2. Try setting the Threshold to where there's about 2dB of compression and notice if there are any words that are less understandable than others. If so, increase to about 6dB and listen again. Don't be afraid to increase the compression, but understand that the more compression you add, the more you'll hear it work and the more color it will add.
3. Set the Attack and Release controls as described previously to breathe with the track, but also experiment with extreme settings.
4. Bypass the compressor and listen to the level, then add makeup gain to make the before and after compression levels the same.
5. With the attack and release set, unsolo the lead vocal and listen to what it sounds like in the track, then tweak as necessary.

***TIP**: Compressing the vocal more may not alleviate the need for automating the track, because even though the vocal may stay at a steady level, the density of the arrangement may change around it.*

Compressing Loops

Much of modern music is derived from samples and loops, but a danger lurks when compressing anything from a loop library, because most of the components may be heavily compressed already. A potential byproduct of using additional compression on a loop is that it could change its groove, which won't be desirable in most cases. That said, sometimes just a few dB of compression can handle any peak that still might exist and allow it to sit better in the mix.

Compression On The Mix Buss

Along with compressing individual tracks, many engineers place a stereo compressor across the mix buss to affect the entire mix as well. Originally the practice came about during the late '70s, when artists began asking why their mixes sounded different in the studio than what they heard on the radio or when their record came from the pressing plant. (This was still back in the vinyl days.) Indeed, both the radio and record did sound different because an additional round or two of compression was added in both mastering and broadcast. To simulate what this would sound like, mixing engineers began to add a touch of compression across the mix buss. The problem was, everybody liked it, so now it's not uncommon for mixes to have at least a few dB of compression added to the stereo mix, despite the fact that it will probably be re-compressed again at mastering and yet again if played on the radio or television.

> *"Compression is like this drug that you can't get enough of. You squish things and it feels great and it sounds exciting, but the next day you come back and you're saying, 'Oh God, it's too much,' so I've been trying to really back it off, especially with stereo buss compression."*
>
> *—Joe Chiccarelli*

When it comes to compressing the mix buss, not all compressors are up to the task. Because only a few dB of compression may be all that's added (although it can be a lot more), the compressor itself actually adds an intangible sonic quality. Among the many favorites are the Fairchild 670 (Figure 9.11) and the Neve 33609 (Figure 9.12).

> *"Generally, the stereo buss itself will go through a Fairchild 670 (serial #7). Sometimes I'll use a Neve 33609 depending on the song. I don't use much, only a dB or 2. There's no rule about it. I'll start with it just on with no threshold, just to hear it."*
>
> *—Don Smith*

Figure 9.11: T-RackS Fairchild 670 plugin

Figure 9.12: Neve 33609
Courtesy Neve

Generally you'll find that most renowned mixers use the buss compressor to add a sort of "glue" to the track so the instruments fit together better, but that doesn't necessarily mean that it requires a great deal of compression. In fact, sometimes only a dB or two of gain reduction at the most is added for the final mix. That being said, many mixers will also offer their clients (artists, band members, producers, and label execs) a more compressed version to simulate what it will sound like after it's mastered. This "client mix" is achieved by using a signal path across the mix buss that's similar to what a mastering engineer would use—that is, a compressor that's fed into a limiter at the end of the chain to raise the level to sound like that of a mastered release (see Figure 9.13).

Figure 9.13: The master buss signal path for a client mix
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Because the clients get used to hearing the "client mix," it's easy for the heavy buss compression to get out of hand. One of the problems with compressing too much is that it leaves the mastering engineer with a lot less room to work, and in the case of a track that's "hyper-compressed," it virtually eliminates the ability for the mastering engineer to be of much help at all. That's why care should be exercised anytime compression is added to the mix buss. (See Chapter 12, "The Master Mix," for more on delivery to mastering.)

That said, there are two times during the mix that you can insert your buss compression: when you first start the mix or toward completion. While the two choices might not seem all that different, you will get a slightly different result from each.

At the End Of The Mix

If you wait to insert the buss compressor until later in the mix, the compressor settings can be less aggressive, because a fair amount of compression may have been already inserted on the individual tracks. If you choose to wait until later in the mix, usually the best time to insert it is at the point where most of your elements and effects have already been added and it's now time to concentrate on final balances.

One advantage of inserting buss compression toward the end is that if you don't like the sound, you can easily substitute a different compressor or even eliminate it all together.

> ***TIP**: If you're using the right compressor, it may take only a dB or two of buss compression to make a sonic difference to where the mix is bigger sounding.*

At the Beginning Of The Mix

The other approach is to insert the buss compressor right at the start of the mix and build your mix into it. Because this affects the dynamics of the mix right from the beginning, mixing this way might take a little getting used to, but it has some advantages.

First, the mix comes together a little quicker since it has that "glue" almost right away as a result. Second, you may find yourself using a little less compression on the individual tracks. This has the secondary benefit of giving you greater control of the overall compression of the mix. If you feel like there's too much, it's easy to back off on the buss compressor to the point where you or your client feel better about the amount, whereas if you added it toward the end, sometimes the only way to dial the total compression of the mix back is to tweak the individual instrument compression, which can take quite a bit of time and rebalancing.

The third thing is that the buss compressor tends to even out the levels of the individual instruments a lot, so you might not need to automate the fader levels as much. The downside of doing it this way is that if you decide you don't like the sound of the compressor, the overall sound and balance of the mix can change a lot when you insert a different one.

Finally, when starting with the buss compressor in the signal path from the beginning of the mix, you may find that you'll be using somewhat more buss compression than if it was introduced toward the end of the mix.

The SSL Mix Buss Compressor

The sound of a great many records from the '80s and '90s comes from the sound of the built-in mix buss compressor on an SSL console (Figure 9.14). This is an aggressive compressor with a very distinct sonic signature. Some have even gone so far as to call the compressor "In" button (meaning it's present in the signal path) the "Good" button, because it makes everything sound better.

Figure 9.14: SSL buss compressor

If you happen to get a chance to work on an SSL console (any vintage—they all have a mix buss compressor), the outboard rack-mount version, the version by Alan Smart, or any of the various plugin emulators, here's the time- honored setting to use as a starting point.

> ***TIP**: The typical SSL buss compressor settings are:*
> - *Attack: All the way slow.*
> - *Release: All the way fast.*

- *Ratio: 4:1.*
- *Threshold: To taste.*

Compression Techniques

Here are a number of techniques often used when adding compression to particular mix elements. Don't limit yourself to just these examples, as the settings can just as easily work for other instruments in other situations as well.

For Snare

- To compress the snare drum to get more sustain so it sounds bigger, go to a part of the song where the drums are playing straight time and adjust the release time so that the sound doesn't begin to attenuate until the next snare hit.

- To remove snare leakage from the overheads without making the hi-hat sound too ambient, compress the overhead mics and key them from the snare by sending the snare signal into the sidechain input of overheads' compressor.

- Instead of adding more high-end EQ to the snare, try compressing it instead, but be sure that the attack is set fairly long so that the initial transient does not get compressed. This will allow you to elongate the snare drum's duration and create the illusion that it is brighter.

> *"What I do a lot is take a snare drum and go through an LA-2, just totally compress it, and then crank up the output so it's totally distorted and edge it in a little bit behind the actual drum. You don't notice the distortion on the track, but it adds a lot of tone in the snare."*
>
> *—Ed Stasium*

- If a sample is being added to the snare (see Chapter 11 for more on how that's done), compress the original snare and not the sample, because the sample usually has fewer dynamics. This works when replacing other drums as well.

For Kick

- For a punchy kick, set the attack time slow enough to let the initial attack through, and set the release time so that it just begins to die at the next kick beat. Increase the ratio to make it punchier sounding.

- For a floppy sounding kick (either with or without a front head), set the release time shorter than normal to deemphasize the end portion of the kick signal.

- Synthesized kicks like from the famous 808 inherently have a lot of low end on them that sounds better when it's controlled. Try some parallel compression by sending it

to another channel that has a compressor inserted, compressing it with a very fast attack and release, and mixing the compressed signal back in with the original to help punch it up.

For Room Mics

- Compress the room mics by 10 to 20dB to increase the room sound with a slow attack time and the release time fairly long or timed to the track. If the acoustics of the room sound good to begin with, it will sound tighter and better than an outboard reverb.

For Bass

- Set the threshold of a compressor to its highest ratio (even if it's infinity:1) with 3 or 4dB of gain reduction. This will keep the bass solid with the level constant in the mix.

- With a bass track that has no definition to the notes, try an 1176 with the attack set to around the middle and the release set to around 3 or 4 o'clock with an 8:1 ratio and a fair amount of gain reduction. This helps put a front end on the notes that may not have been articulated properly when the part was recorded.

- With a bass track where the notes don't sustain long enough, increase the release to where the longest notes don't die until the next one sounds.

- If a bass player is using a pick, it may create high midrange transients that might need to be limited even before compression is added. Insert a limiter first and set the attack and release times fairly fast, then insert a compressor set from medium to slow on both attack and release and a ratio of 4:1.

- On some rock bass parts played with a pick, a multiband compressor across the bass can achieve a more even level without sounding compressed.

For Vocal

- A good starting point for a lead vocal is a 4:1 ratio, medium to fast attack and medium release, and the threshold set for about 4 to 6dB of gain reduction.

- When a single compressor or limiter just isn't enough for a troublesome vocal, try an 1176 (hardware or plug- in) set to fast attack (all the way clockwise, which is backward from other compressors), the release set to medium (5), and the ratio set to either 8 or 12:1 to clip the peaks by 4 to 5dB. Feed the output of the 1176 into an LA-2 (again, hardware or software) set for gentle gain riding of 2 to 3dB.

- As an option to above, if you have access to another 1176 or have enough DSP to insert a second, set the first one on fastest attack and an 8:1 ratio, and the second on a

slower attack and a 4:1 ratio. With both set for about 4 to 6dB of compression, the vocal will be silky and smooth, yet in your face.

For Piano

- If you liked the early Elton John piano sound, put the piano into two LA-2As or similar compressors (or stereo versions of the plugin) and compress the signal a large amount (at least 10dB), then connect the output into two Pultecs or similar equalizers. Add 14kHz and 100Hz to taste. The effect should be a shimmering sound where the chords hold and seem to chorus.

For Guitar

- Higher ratios of compression around 8 or 10:1 sometimes work well, with the threshold set so that the guitar cuts through the track. Attack and release should be timed to the pulse of the song.

> *"I may go 20:1 on a [UREI] 1176 with 20dB of compression on a guitar part as an effect. In general, if it's well recorded, I'll do it just lightly for peaks here and there."*
>
> *—Don Smith*

- When a guitar is recorded with both close and distant mic tracks, you can use this trick to get guitars to sound big, yet stay out of the way of the vocal. Pan the distance mic in the same direction of the close mic and then attach a compressor across the distant track keyed off the lead vocal. When the lead vocal is present, the ambience is decreased. When the lead vocal stops, the ambience returns to full level. As a variation, try panning the distance mic track to the opposite side of the close mic track.

- With doubled guitars, pan them medium left and right, then send the same amount of both to a compressor on a separate track, either via an aux send or a buss. Pan the output of the compressor track to the center and bring it up just underneath the original tracks. Now the guitars don't have to be as loud as before to still have presence.

- If you notice a guitar getting lost in places, try compressing with a ratio of 2:1 to 4:1 with medium attack and release times. This will allow the rhythmic transients to get through somewhat uncompressed while boosting the sustain of the guitar sound.

- With rock guitars, the idea is to have them big and "in your face." This is accomplished by first limiting the transients from the signal with a ratio of 10:1 or higher. Be careful to set the release so that any sustaining parts of the signal return to

unity gain before the next transient.

Using A De-Esser

Sibilance is a short burst of high-frequency energy where the S's are overemphasized, which comes from a combination of mike technique by the vocalist, the type of mic used, and heavy compression on the vocal track. Sibilance is generally felt to be highly undesirable, so a special type of compressor called a de-esser is used to suppress it (see Figure 9.15).

Figure 9.15: A typical de-esser
Courtesy Empirical Labs

Most de-essers have two main controls, Threshold and Frequency, which are used to compress only a very narrow band of frequencies anywhere between 3kHz and 10kHz to eliminate sibilance. Modern software de-essers are much more sophisticated, but the bulk of the setup still revolves around those two parameters. One frequently used additional feature is a *Listen* button that allows you to solo only the frequencies that are being compressed, which can be helpful in finding the exact band of offending frequencies.

To use a de-esser, do the following:

1. Insert the de-esser on the vocal channel and solo it.
2. Lower the Threshold control until the sibilance is decreased, but you can still hear the S's. If you can't hear them, then you've raised the Threshold too far.

3. Scan the available frequencies with the Frequency control until you find the exact spot where it's most offensive, then adjust the Threshold control until the S's sound more natural.
4. Un-solo the vocal and listen in context with the track. Be careful that you don't completely eliminate all the S's, because that makes it sound unnatural.

***TIP**: When using the Listen feature, remember that the audio you're hearing isn't in the signal path, just the sidechain. Don't forget to disengage Listen when you've found the correct frequencies.*

Using A Gate

Like the de-esser, a gate can sometimes consist of just a few controls, principally the Threshold, Range, and sometimes a Hold or Release control (see Figure 9.16). Range sets the amount of attenuation after the threshold is reached and the gate turns on.

Sometimes when gating drums, the Range control is set to attenuate the signal only about 10 or 20dB. This lets some of the natural ambience remain and prevents the drums from sounding choked. The Hold control keeps the gate open for a defined amount of time, and the Release control sets how quickly the gate closes again.

Figure 9.16: A typical noise gate

We'll use the snare drum as an example of how to set up a gate, but the same technique is used to set up a gate on any drum, instrument, or vocal.

1. Insert the gate on the snare channel and solo it.
2. Raise the Threshold control until you can hear the snare drum hit, but there's no sound in between the hits.
3. If the snare sounds unnatural or cut off, raise the Threshold a bit to see whether that improves the sound.
4. If the snare still sounds unnatural or cut off, try increasing either the Hold or the Release control.
5. Try adjusting the Range control so the snare is attenuated by 10 dB between hits to hear whether it sounds more natural.
6. If the gate chatters, try fine-tuning the settings of the Threshold and Release controls (if the gate has one or both). This may require some experimenting, so be patient.
7. Try timing the Release control so that the gate breathes with the pulse of the song.

***TIP**: Gates are finicky because the dynamics of the signal are usually constantly*

changing. Inserting a compressor in the signal chain before the gate improves the performance greatly.

Gating Techniques

Here are some techniques often used when gating a mix element. Gates usually take time to set up properly, so take your time and don't be afraid to experiment.

For Snare

- This simple technique allows a different effect to be placed on the snare during harder hits and prevents leakage to the effect during tom hits, hi-hat, or stray kick-drum beats. Duplicate the snare to another channel and insert a gate on this new channel. This gated channel is generally not sent to the main mix (although it can be), but it is used primarily as an effects send. By adjusting the threshold, you can have more control over how the signal is sent to the effects unit.

- Another way to make the snare feel as if it's breathing with the track is to copy the snare onto a second track and insert a gate on only that channel. Time the release so it cuts off right before the next snare hit during the main part of the song when the snare is just playing time. Bring the gated snare channel up just underneath the main snare track.

- To make the snare drum sound bigger, gate either room ambience or reverb and trigger it from the snare by sending the snare signal to the trigger/key input of the gate.

For Drums

- When gating toms, set the Range control so it attenuates the signal only about 10 or 20dB. This lets some of the natural ambience remain and prevents the drums from sounding choked or unnatural.

- To make the rhythm section feel tighter, feed the bass through a gate with only 2 or 3dB of attenuation when the gate is closed. Trigger the gate from the kick drum so that the bass is slightly louder on each kick drum beat.

- For times when the groove doesn't quite lock or the bass player is playing on top of the beat and the drummer is laying back, insert a gate on the bass channel and key it from the kick drum. The bass will only play when the kick drum is played, so the rhythm section will now sound tight. Set the sustain and release so that it sounds more natural and less staccato, since the kick is a transient instrument and the bass is not, or leave it staccato if that works well.

- The above technique can also be used to get more space and rhythm in a big chorus by gating the rhythm guitar track keyed from the hi-hat.

- Key a gate placed across a synth or keyboard pad from the hi-hat to make the pad

more rhythmic.

- For tighter background vocals, patch the best phrased harmony line into the key insert of a gate across a stereo submix of the harmonies. This will ensure that all of the vocal entrances and exits will remain tight.

Chapter 10

The Interest Element: *The Key To Great (As Opposed To Merely Good) Mixes*

Although being skilled in applying the previous five elements may be sufficient for many types of audio mixes, most hits require a mix that's taken to another level. Although it's always easier with great tracks, a solid arrangement, and spectacular playing, a great mix can take simply okay tracks and transform them into something so compelling that people can't get enough of it. It's been done on some of your all-time favorite songs.

So how can we get a mix to that point?

More than being just technically correct, a mix must be as interesting as a good movie. It must build to a climax while having points of tension and release to keep the listener subconsciously involved. Just as a film looks bigger than life, a great mix must sound bigger than real life. The passion and the emotion must be on a level where the listener is sucked in and forced to listen.

Which brings us back to where we started:

- **Determine the direction of the song**
- **Develop the groove**
- Find the most important element and emphasize it

> *"The tough part, and the last stage of the mix, is the several hours it takes for me to make it sound emotional and urgent and exciting so that it's not just a song, it's a record. It's not making it just sound good, it's making it sound like an event. That's the last stage of when I mix, and that's the part that makes it different or special."*
>
> *—Ed Seay*

The Direction Of The Song

The first thing that the mixer does before diving headfirst into the mix is find the direction of the song, and that's determined by both the style of the artist and the performances by the players. For instance, if the song is folksy in nature, then it probably won't need big, bombastic drums and long reverbs and delays, but if the artist is a loud arena rock band, then you probably won't want a close, intimate sound.

It's absolutely possible to change the direction of the song and have a hit, although it's usually based on the style of the artist. A good example of this is Marvin Gaye's classic "Heard It through the Grapevine," which has been a hit by many artists in innumerable styles through the years. The direction of Creedence Clearwater is very different from the direction of Gladys Knight and the Pips, the Kaiser Chiefs, or Amy Winehouse, yet it works equally well for all of them. The direction is a function of the artist and the performance.

Develop The Groove

All good music, regardless of whether it's rock, jazz, classical, rap, marching band, or some new space music that we haven't heard yet, has a strong groove. So what exactly is the groove?

The groove is the pulse of the song.

A common misconception about a groove is that it must have perfect time. A groove is created by tension against even time. That means that it doesn't have to be perfect, just even, and that all of the performances don't have to have the same amount of "even-ness." In fact, it makes the groove stiff-feeling if the performances are too perfect. This is why perfect quantization of parts and lining every hit up to a grid in a DAW frequently takes the life out of a song. It's too perfect because there's no tension. It's lost its groove.

Just about every hit record has a great groove and that's why it's a hit, but if you really want to study what a groove is, pick any song from one of the masters: James Brown, Sly Stone, George Clinton, or Prince. Every song contains the audible essence and feel of a groove.

We usually think of the groove as coming from the rhythm section, especially the drums, but that's not necessarily always the case. In the Police's "Every Breath You Take," the rhythm guitar establishes the groove, while in many songs from Motown's golden age by the Supremes, the Temptations, Stevie Wonder, and the Four Tops, the groove was established by James Jamerson's or Bob Babbitt's bass. It's even been said that Michael Jackson's vocal was a groove unto itself and a song could easily be built around it.

The trick for the mixer is to find what instrument best defines the song's groove and then build the rest of the mix around it.

Finding The Groove

Regardless of which instrument is providing the groove of the song, if you want a great mix, you've got to find it before you do anything else. Have a listen to a rough mix of the song or just push all the faders up and see whether you can feel its pulse.

Ask yourself the following:

1. What instrument (or instruments) is providing the pulse? Is it the drums? The bass? A keyboard? A loop?
2. What would make the groove stand out? The balance? The tone? Compression?
3. Is there a rhythm arrangement element to the mix, such as percussion, a rhythm guitar, a keyboard, or a loop (check back to Chapter 5)? Is it essential to the groove?

Building The Groove

While it's true that sometimes the groove may be the result of a single instrumental performance, usually it's built around the interplay of a number of instruments, especially in complex mixes with a lot of tracks.

Normally, the groove of the song is provided by the bass and drums, but it's important to determine whether another instrument, such as a rhythm guitar, keyboard, loop, or percussion, is an integral part that makes up the pulse of the song. Usually this can be easily identified as an instrument that's playing either the same rhythmic figure as the bass and drums or a multiple of the rhythm, such as double time or half time.

After those additional rhythmic elements are discovered, here's one way to build the groove:

1. Find the instrument that provides the basic pulse of the song (such as the drums).
2. Add the lowest-frequency instrument that's playing the same or a similar rhythmic figure (usually the bass).
3. Add any additional instruments playing the same or a similar rhythmic figure in order of frequency from low to high.
4. Add any instrument playing a similar rhythmic figure, such as half or double time.
5. Add any instrument working as the rhythm arrangement element (remember the section about arrangement elements in Chapter 5?) and providing motion to the song (such as a shaker or tambourine).

The groove may be attributed to only a single instrument, such as in the case of a power trio (guitar, bass, and drums), to three or even four instruments on rare occasions. If you're not sure, the best way to determine what's playing the groove is to try mixing in different combinations of instruments along with the rhythm section to see whether the pulse gets stronger, gets weaker, or stays the same.

> ***TIP**: If a new instrument adds to the pulse of the song and the pulse seems lessened if it's muted, then you have an instrument that's a big part of the groove.*

Find The Most Important Element And Emphasize It

Equally as meaningful, and in some cases even more important than the groove, is finding whatever element is the most important to the song. In some cases (such as EDM, dance, and hip-hop music), the most important element is the groove, yet in other genres it may be the vocal (such as in country or pop).

Even though the most important element is often the lead vocal, it doesn't necessarily have to be. It could be a riff, such as the intros to the Stones' "Satisfaction" and "Start Me Up," the piano line from Coldplay's "Clocks," or the guitar line from the White Stripes' "Seven Nation Army." It's always a part so compelling that it forces you to listen and re-listen to the song.

Whatever part is most important, the mixer must identify and emphasize it in the mix in order for the mix to be elevated beyond the ordinary. Like most other creative work, that requires inspiration, but you can't underestimate the value of talent and experience in the process.

To emphasize the most interesting element:

1. After the element is determined, raise its level a dB and then increase in 1dB steps. Does the part jump out of the mix?
2. If the part isn't compressed, insert a compressor and begin by adding 2dB of compression. If the part is already compressed, add an additional dB and increase in 1dB steps. Be sure to compensate for the decrease in level by adjusting the Output control. Does the part jump out of the mix?
3. If the part is already EQ'd, add an additional dB at the EQ points previously selected. If the part isn't EQ'd, insert an equalizer and add 1dB at 5kHz. Does the part jump out of the mix now? What if you add a dB at 1kHz?
4. If the part is dry, try adding some effects, starting with any reverb or delay that's already set up and in use in the mix. If that doesn't work, add a dedicated reverb beginning with the parameters set from short to long. Move on to delay, modulation, distortion and any other effects at hand. Does the part jump out of the mix now?

> *"I try to find what's important in the mix. I try to find out if the lead vocal is incredibly passionate and then make sure that the spotlight shines on that. If, for instance, the mix needs 8th notes, but they're going [makes sound effect] and it's not really pushing the mix, sometimes playing with compression on the acoustics [guitars] or auditioning different kinds of compression to make it sound like, 'Boy, this guy was into it.' It's just basically playing with it and trying to put into it that undefinable thing that makes it exciting. Sometimes hearing the guy*

breathe like the old Steve Miller records did. With a little of that, you might say, 'Man, he's working. I believe it.' It's a little subconscious thing, but sometimes that can help."

—Ed Seay

Fifteen Steps To A Better Mix

Mixing is a nebulous art in that most musicians and engineers learn more by feel and listening than by being taught. As a result, a number of important items in a mix can be easily overlooked, and these can mean the difference between a mix that sounds polished and professional and one that sounds amateurish.

Here's a checklist of items that can help you think in a little more detail about your mix and tighten it up as a result.

1. **Does your mix have dynamic contrast?** Does it build as the song goes along? Are different instruments, sounds, or lines added or muted in different sections?
2. **Does your mix have a focal point?** Is the mix centered around the instrument or vocal that's the most important?
3. **Does your mix sound noisy?** Have you eliminated any count-offs, guitar amp noises, bad edits, and breaths that stand out? Each one may not seem like much, but their effect is cumulative.
4. **Does your mix lack clarity or punch?** Can you distinguish every instrument? Does the rhythm section sound great by itself? Is the balance between bass, kick, and snare correct?
5. **Does your mix sound distant?** Try decreasing the level of the reverb and effects, starting with the wettest and then working your way to the least wet.
6. **Can your hear every lyric?** Every word must be heard. Tweak the automation if you're using it; automate the track if you're not.
7. **Can you hear every note being played?** If solos or signature lines are being masked, automate the track to hear every note, or tweak the automation until you can.
8. **Are the sounds dull or uninteresting?** Are generic synth patches or predictable guitar or keyboard sounds being used? Try modifying them with an effect.
9. **Does the song groove?** Does it feel as good as your favorite song? Is the instrument or instruments that supplies the groove loud enough?
10. **What's the direction of the song?** Should it be close and intimate or big and loud? If your current direction isn't working, try the opposite.
11. **Are you compressing too much?** Does the mix feel squashed? Is it fatiguing to listen to? Is all the life gone? Decrease the mix buss compression first. Decrease the main instrument or vocal compression next. Decrease the rhythm-section compression next. Decrease the compression on everything else lastly.
12. **Are you EQing too much?** Is it too bright or too big? Decrease the upper midrange EQ on the vocals, guitars, loops, and snare. Decrease the low-frequency EQ on the bass and

kick.

13. **Are your fades too tight?** Does the beginning or ending of the song sound clipped? Adjust the fades until the attack transients of the notes are distinct.
14. **Did you do alternate mixes?** Did you do at least an instrumental-only mix?
15. **Did you document the keeper mixes?** Are all files properly named? Are you sure which file is the master? Have you made a backup?

An interesting mix is all in the details, and those take time to sort out. Working through each one of these steps may take a while, but the end result can definitely be worth it.

Chapter 11

Advanced Techniques

Whereas a decade ago a mix was deemed complete as soon as it felt good to everyone involved, mixes these days require more precision than ever before. This is because the mindset of a mixer has changed thanks to the ability to now perform many more mix moves in a repeatable manner inside a DAW. In this chapter we'll look at a few of the more advanced techniques used to make a mix competitive today. Keep in mind that cleanup, timing adjustment, and pitch correction are considered more a part of production and should be completed before the track is sent to be mixed, but mixers are sometimes expected to perform these tasks anyway.

Cleanup

As stated in Chapter 3, a big part of today's mixing can be track cleanup. Although it's always preferable for this to take place prior to mixing, sometimes the previous engineers don't have time, are careless, or maybe even don't know how to properly clean up the tracks before mixing begins.

What's meant by cleanup? For the tracks to be as clean as possible with the least amount of aural distraction, it's important that any clicks or pops are removed, the bad fades are fixed, and any noise is eliminated. Here's how to do it.

Removing Noise

Noise on a track can mean anything from the guitar-amp buzz before the guitarist plays a part, to the shuffling of feet or the clearing of the throat captured by the vocal mic before the singer begins to sing. As a matter of course, the length of each audio clip should be trimmed to just before and after the part plays to eliminate the noise (see Figure 11.1). Be sure to add a short fade-in or fade-out to eliminate any potential clicks (see Figure 11.2).

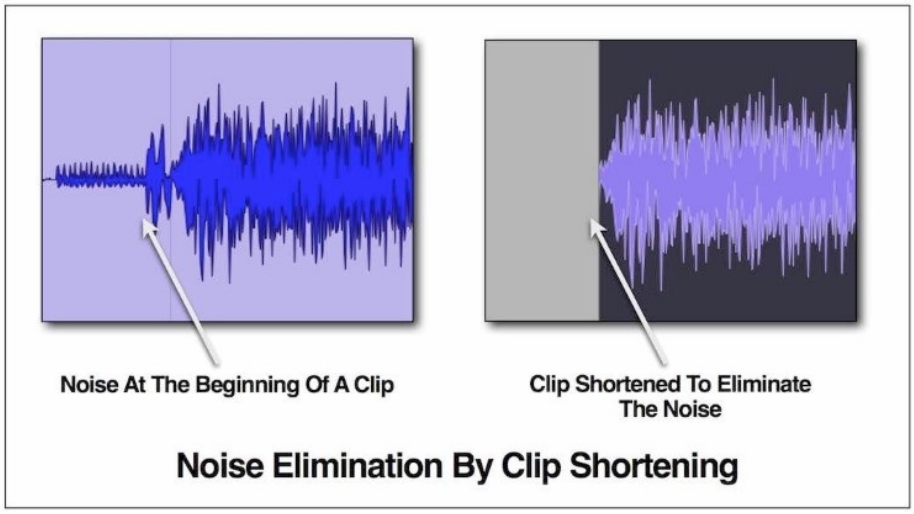

Figure 11.1: Shortening the clip length to eliminate noise
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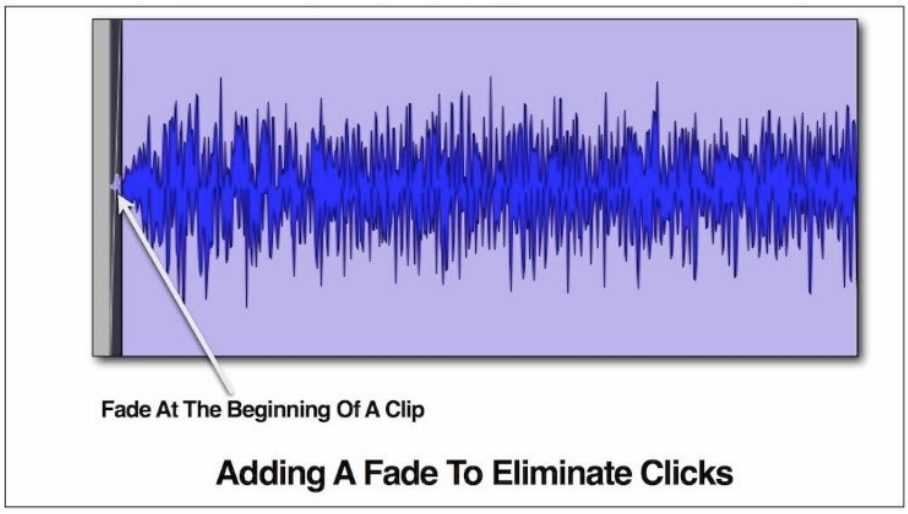

Figure 11.2: Adding a short fade to eliminate clicks
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While shortening the clip doesn't usually pertain to the drum track because doing so can make it sound choppy and unnatural, there are times when it's appropriate, such as in the case where there's a measure or more of silence in the song. Be sure to use a fade that both sounds natural and doesn't cut off the cymbal decay or the attack of the next downbeat after the silence (see Figure 11.3).

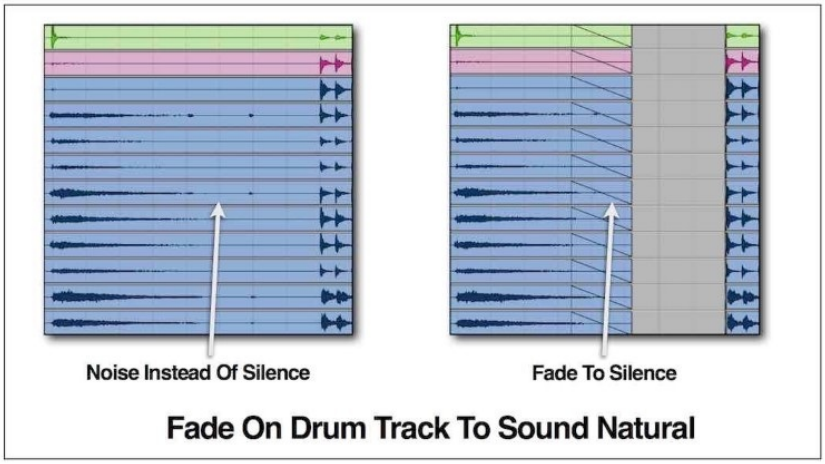

Figure 11.3: Adding fades to a drum track
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Removing Clicks And Pops

Clicks and pops can come from a number of situations, including butt-cut edits (edits without fades that are butted up against one another) or transient overloads, such as from a thumb-slapped bass with active pickups. Most clicks occur during cut and paste, timing adjustment, or noise elimination operations where a fade isn't used so only a butt- cut remains (see Figure 11.4). Although sometimes the click is very apparent, many times it's quiet enough that it will get lost in the track when the other instruments are introduced. Sometimes an edit can be completely silent if it happens to be made in the right place in the audio waveform.

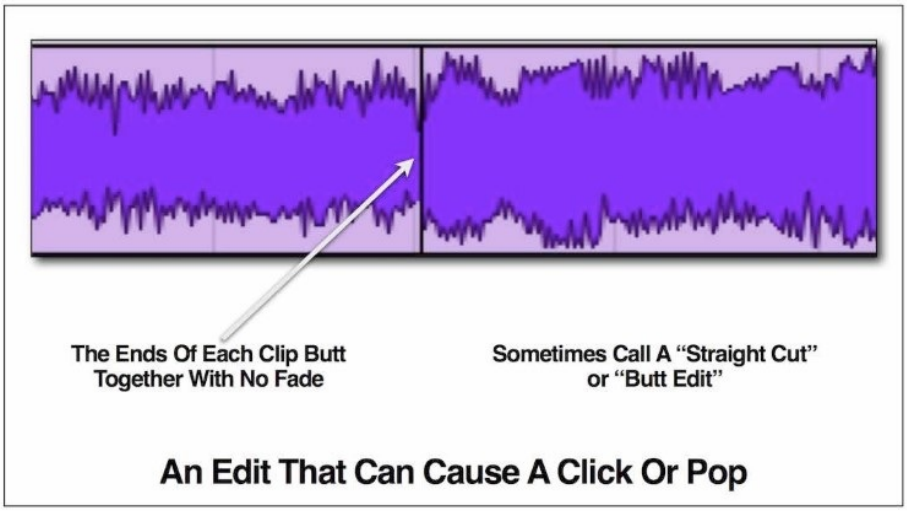

Figure 11.4: A butt-cut edit
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> ***TIP**: While you may not hear the clicks when the track is played back over monitors, you may find that they jump right out when you're listening with headphones. That's why it's best to always listen to a pass with phones to see whether any clicks are audible.*

The way to fix any potential trouble spots is by adding fades to the edits. This is something that you can do by eye, although it's time consuming. While you may hear clicks on butt-cuts at the beginning and ending of clips, it's more likely that the ones that will stick out are edits that lack crossfades. Go through the track and add crossfades where appropriate (see Figure 11.5).

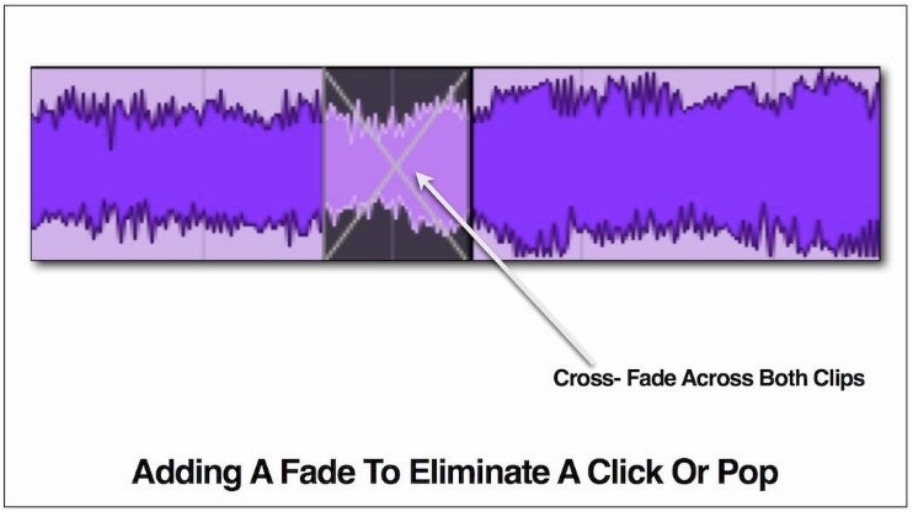

Figure 11.5: Adding a crossfade
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Remember that even though these clicks might not be readily apparent, any kind of noise accumulates and plays a very subtle role in how the listener perceives the song. Sometimes he can’t explain the difference, but he can hear it.

Removing Count-Offs

Leaving count-offs (some musicians call them “count-ins”), such as drum stick clicks, is a sure sign of a demo recording and is something that no one wants to listen to. While you can set any DAW to begin precisely on the downbeat of the song after the count-off, it’s best to eliminate the count-off altogether, especially if there’s another instrument that’s beginning the song against what’s supposed to be silence.

There are two ways to eliminate a count-off; either shorten the clip length as we did previously for noise elimination, or make a cut so there’s a separate clip containing the count and then mute it (see Figure 11.6). Muting the clip leaves the count intact in the event that the count is needed later for additional overdubs (although this shouldn’t be needed again once it’s time to mix).

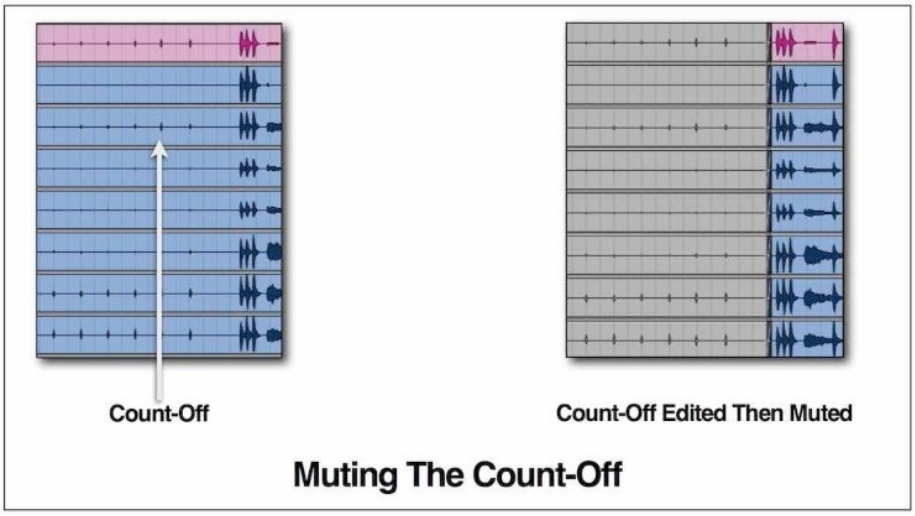

Figure 11.6: Creating a separate count clip and muting it
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Fixing Bad Fades

Sometimes adding a default fade just doesn't sound natural. Either the fade is too tight and cuts off the attack or release of the part, or the fade itself just isn't smooth sounding enough. Now is the time to fix any fades that don't work in the track by adjusting the fade timings (see Figure 11.7).

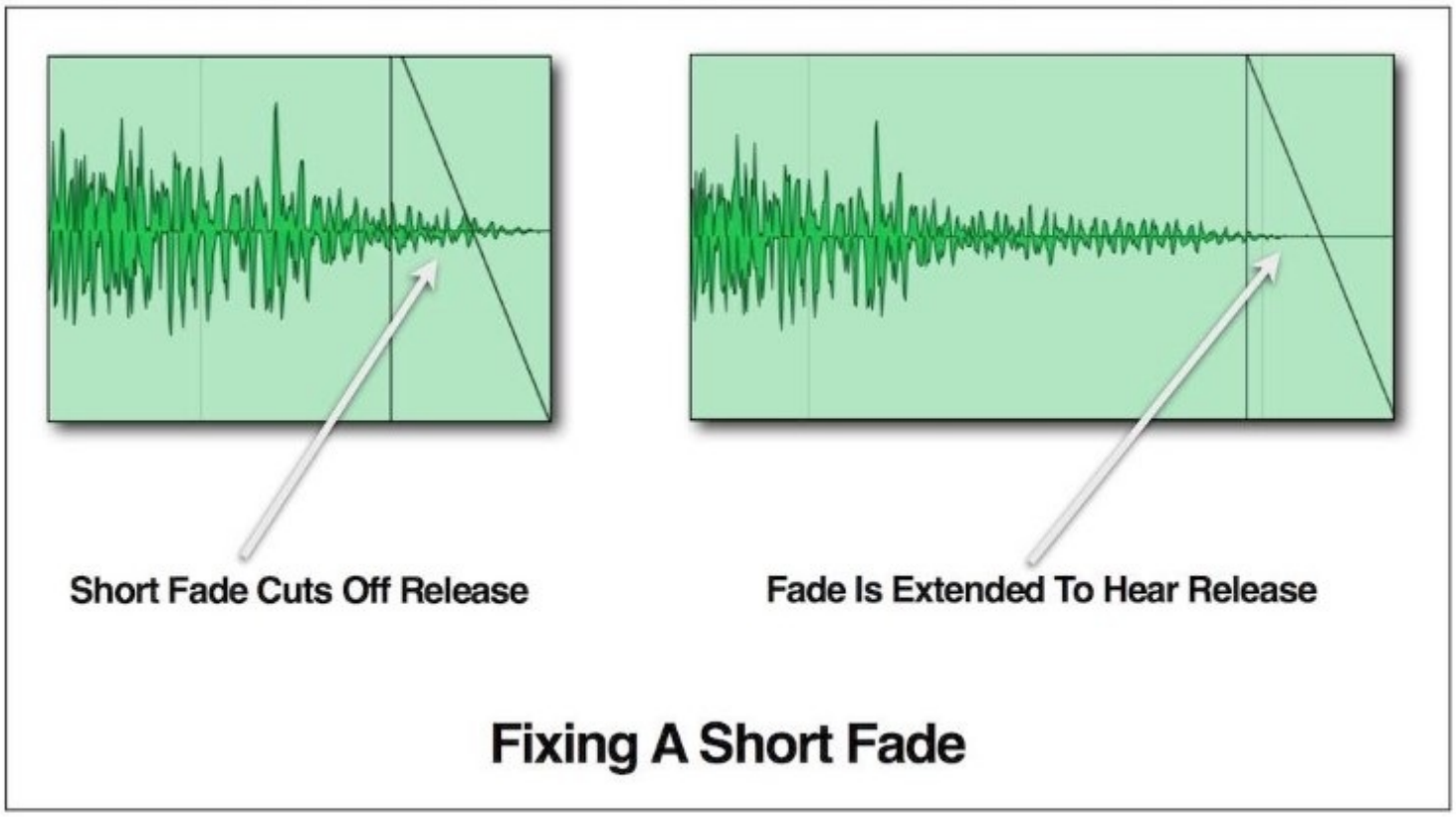

Figure 11.7: Fixing a short fade
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In the case of a fade that seems unnatural (especially a fade-out), try an exponential power fade instead of the default fade (see Figure 11.8).

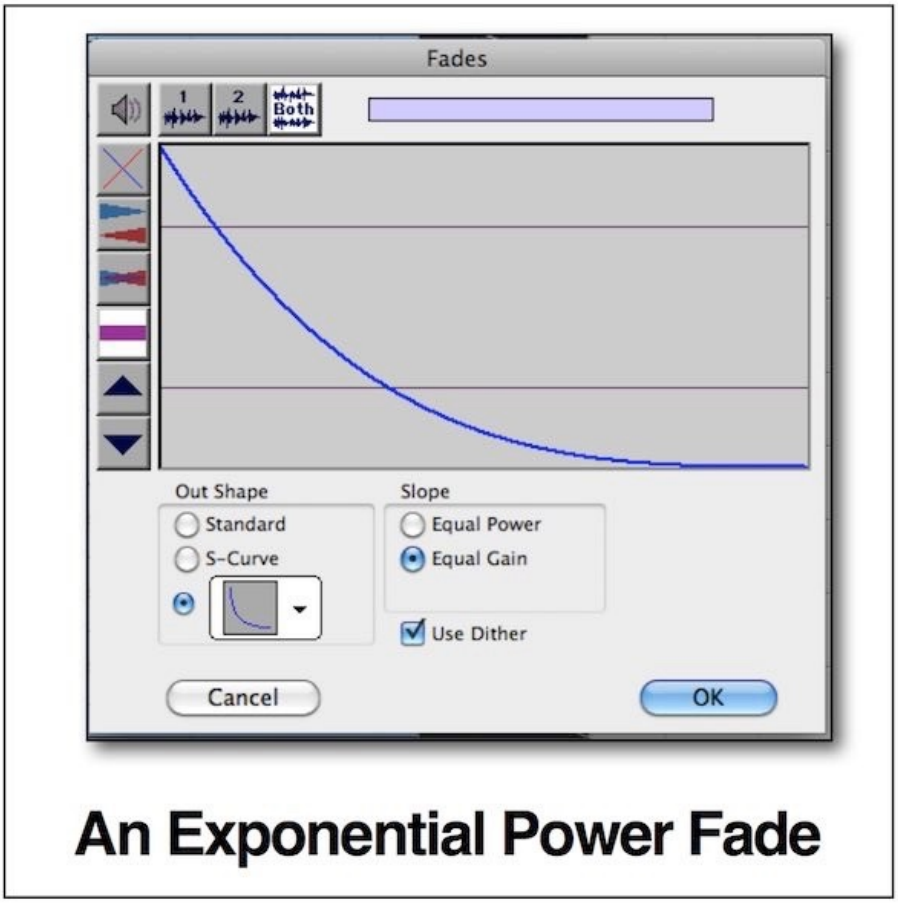

Figure 11.8: An exponential power fade
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Eliminating Unwanted Distortion

Distortion can be anything from a breath blast to a note on a direct bass that's been clipped. In both cases it's the transient of the attack that causes the problem, but thankfully it's extremely short, and there are a number of ways to either fix it or mask it.

Replacement

Sometimes the most natural sounding operation is to replace the area of distortion with a similar area from another part of the song that's clean. That means finding a similar piece, copying it, and then pasting it over the section of the clip that has the distortion or noise (see Figure 11.9).

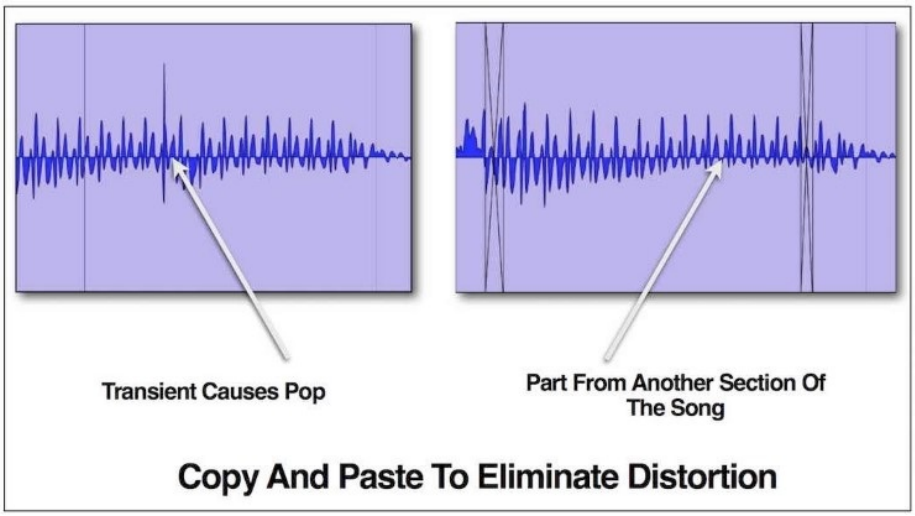

Figure 11.9: Copy and pasting to eliminate distortion
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Clip Level Adjustment

If a clean section of the song isn't available to copy and paste, the next thing to try is to adjust the level of just the transient. Make a cut around the transient and lower the level until it sounds natural (see Figure 11.10), provided your DAW has a clip based level feature.

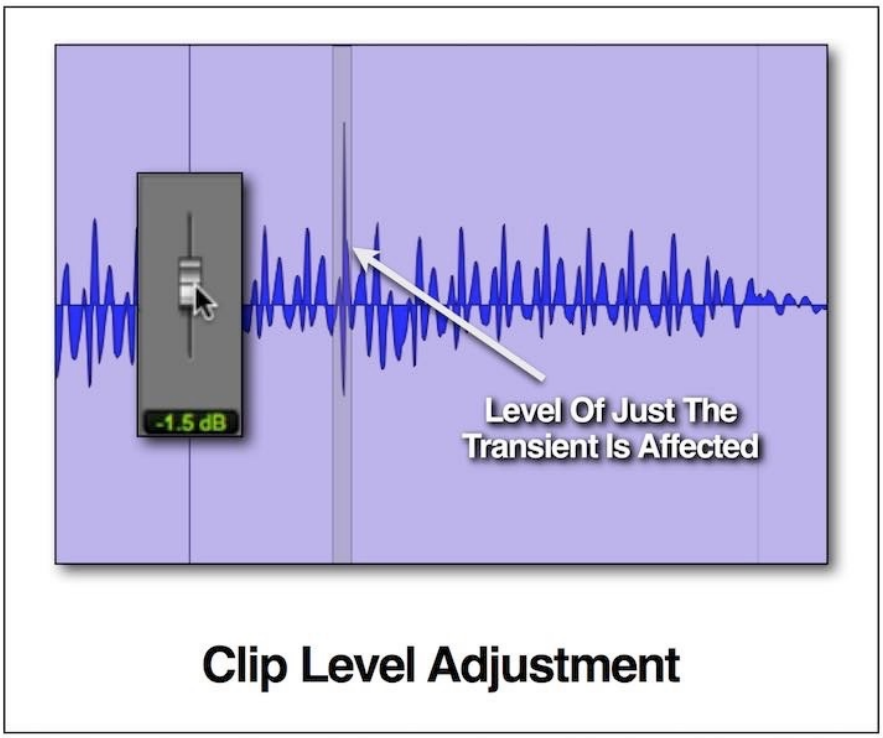

Figure 11.10: Treating the transient with a clip level adjustment
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Automation

If a clip level adjustment feature isn't available on your DAW, you can accomplish the same thing using track automation. Draw the automation curve so that the level of the transient decreases to a point where it seems natural (see Figure 11.11). You can read more about using automation later in this chapter.

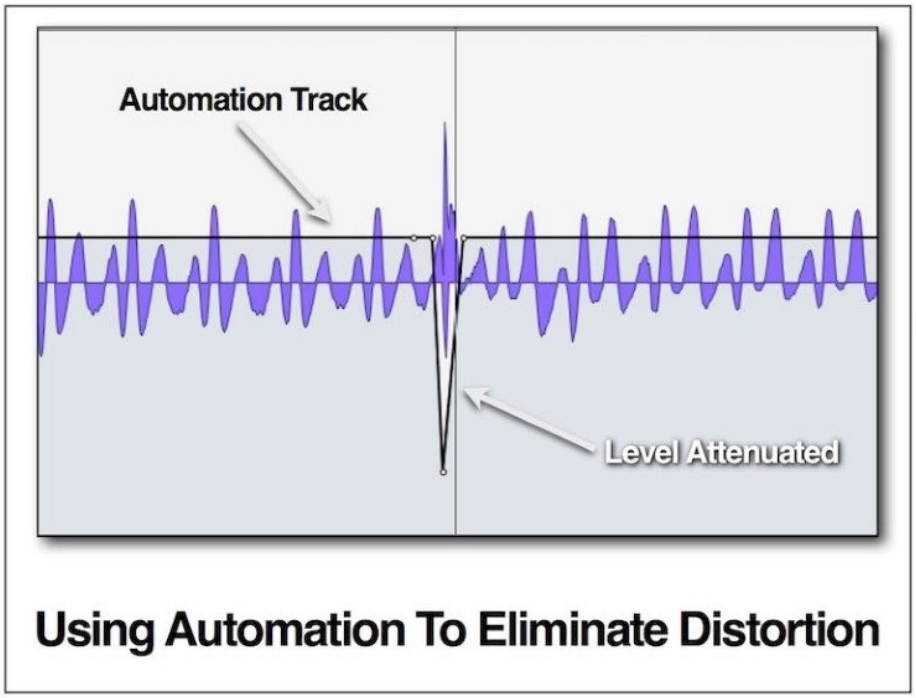

Figure 11.11: Using the automation to eliminate distortion
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Elimination

Sometimes a transient passes so quickly that you can easily delete it without it being noticed. Be sure to add fades to the new edits so as to not create a new click (see Figure 11.12).

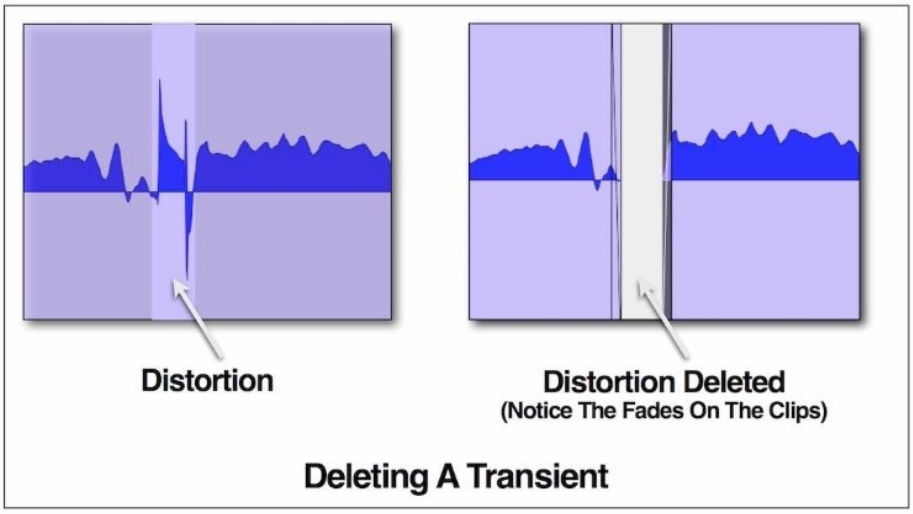

Figure 11.12: Deleting a transient
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Deleting Extra MIDI Notes

Delete any extra "split" notes that were mistakenly played when the part was originally recorded. You might not hear all the notes play distinctly when all the instruments of the track are playing together, but just like the noise at the beginning of tracks, they have a tendency to come to the forefront after things get compressed (see Figure 11.13) and are cumulative in nature, just like noise.

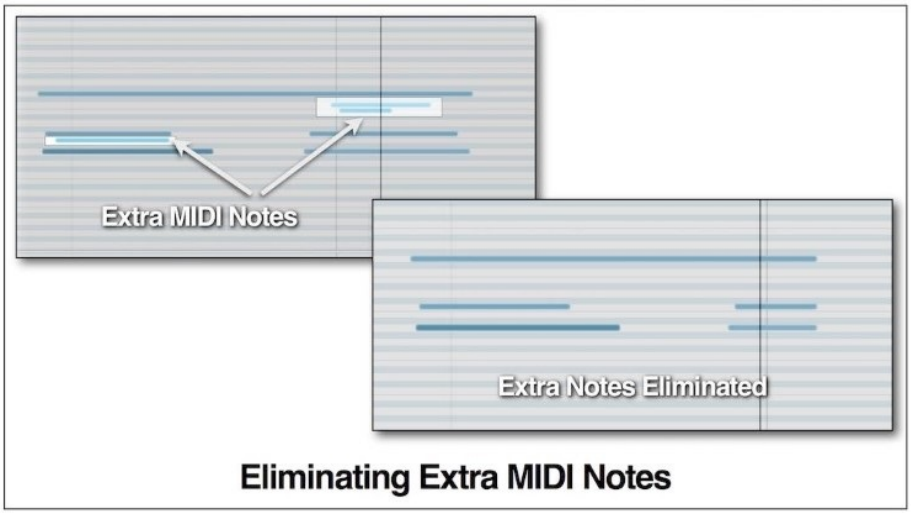

Figure 11.13: Deleting extra MIDI notes
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Adjust The Timing

No matter how great the players on the session are, there's always some portion of a recording that doesn't feel quite right. Usually, the timing of the basic tracks will be tweaked right after your tracking session so you have a solid rhythm section to overdub against, but if you're just now discovering some sections of an overdub that don't feel right (which happens more than you might think), prepare for the joys of slipping and sliding time.

Here's a list of some of the dos and don'ts for tweaking the track timing:

- **Don't edit by eye.** In most music (electronic music being the exception), you can't successfully edit by just trying to line everything up to the kick and snare or the grid and still have it sound natural and human. Often, tracks that look perfectly lined up don't sound or feel right, which is why listening is more important than looking. Turn your head away from the monitor or close your eyes and just listen before and after you move anything.

- **Every beat doesn't have to be perfect.** In fact, if it's too perfect, you'll suck the life out of the performance. Unless something really jumps out as being out of time, you might get away with leaving it as is. Another way is to just line up downbeats and any major accents, which gives you the best of both worlds: a loose feel that still sounds tight.

- **Copy and paste another section.** If you have to make too many edits to a particular section, chances are it won't sound as good when you're finished as just finding a similar section in another part of the song and pasting it in over the area that's suspect. It's a lot faster and easier to do and will probably sound cleaner and groove better as well.

***TIP**: Many times the bass will speak better if it's a few milliseconds behind the kick drum rather than right with it. It still sounds tight, but both the kick and bass will be more distinct, and the sound may even be fuller.*

- **Listen against the drums.** If you listen to the track that you're editing all by itself, you can be fooled into thinking that the timing is correct when it's not, especially if you're editing to a grid. The real proof is when you listen against the drums. If the instrument sounds great by itself and great with the drums, you're home free.

- **Trim the releases.** This is one of the best things you can do to tighten up a track.

Everyone is hip to tightening up the attacks, but it's the releases that really make the difference. Regardless of whether it's an accent played by the full band, the song ending, or a vocal or guitar phrase, make sure that the releases are pretty much the same length. If one is longer than the rest,

trim it back and fade it so it sounds natural. If one is a lot shorter than the rest, use a time correction plugin to lengthen it a bit (see Figure 11.14).

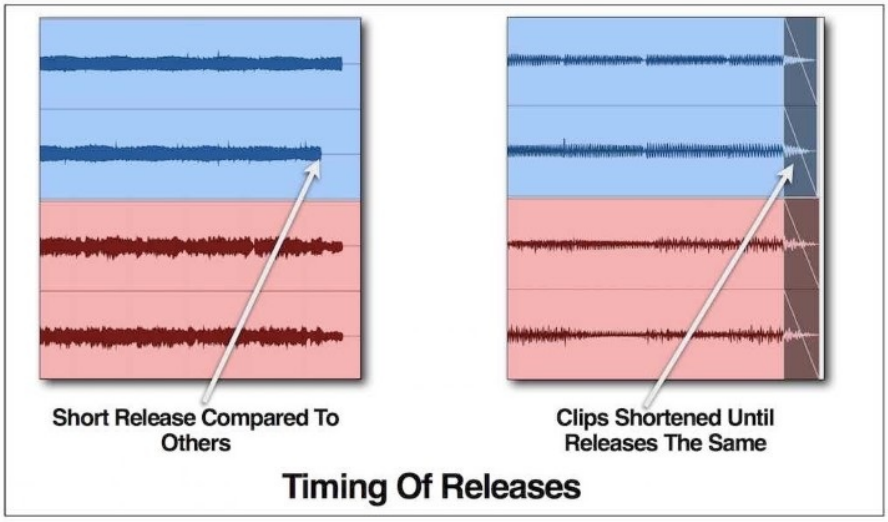

Figure 11.14: Timing of releases
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Of course, if you're using loops or MIDI instruments, you've probably quantized things to the track by now. If you haven't, remember that if it's too perfect to the grid, it may not sound natural.

Pitch Correction

Depending upon how much of a purist you are, pitch correction is either the worst thing to ever happen or a godsend. Regardless of how you come down on the issue, it's at the very least a necessary evil in today's music.

Tuning vocals has been done since way back in the early '70s, starting with the first Eventide H910 Harmonizer (see Figure 11.15). Primitive as it was, it did allow for slight pitch changes, although the digital artifacts that it imposed on the sound were quite substantial.

With every new model of pitch shift hardware that was subsequently introduced, the technology improved to the point where today there are some excellent plugins commonly available that would simply astound any engineer transported in time from back then.

Figure 11.15: An Eventide H910 Harmonizer
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There are three popular track tuning programs commonly in use: Antares Auto-Tune, Waves Waves-Tune, and Celemony Melodyne, as well as less often used variations, such as Avid's Elastic Audio or Cubase VariAudio. Be aware that all tuning programs impart their own sound on the audio you're tuning, and it might not always be pleasing, which is why many engineers own several different ones so they can compare which one sounds better in a particular situation.

Although the way pitch correction plugins are used is somewhat the same, there are a number of guidelines that are worth following:

- **Use the performance itself first.** Before you apply pitch correction, exhaust all other remedies to keep the performance as natural sounding as possible. These include vocal comping and copy and pasting phrases, words, or syllables from other parts of the song.
- **A little goes a long way.** The fewer notes you correct, the more natural the performance will sound. You're much better off just correcting a few notes than attempting to correct the entire performance.

***TIP**: Generally, background vocals can get away with much more pitch correction than lead vocals without being heard.*

- **Use the most precise mode.** Most engineers avoid using Auto mode because it's not precise enough for most applications, and as a result, it causes audible fluctuations that make the vocal sound like it's tuned, which is usually not what you're after (Cher and T-Pain aside). Use the graphical mode if the plugin has one in order to achieve the most precise tuning with the least amount of audible artifacts.

- **Don't perfectly tune the vocals.** Even the best vocalists are never precisely on pitch, so if you tune it that way, it may sound unnatural. Getting the pitch within a few cents will sound more like the real thing, since it's the variations and inaccuracies that make a human voice sound human.

> *"Of the four albums I co-produced with David Bowie, 95 percent of all his vocals were only one take. There are small errors in pitch and timing if you listen closely, but that's what makes David sound like David."*
>
> *—Ken Scott*

- **Print the pitch correction.** Instead of leaving the pitch correction patched in as a plugin, you're better off to print a corrected pass and use that vocal instead. This not only saves precious system resources, but it also eliminates any problems that might occur if the session is moved to a DAW with a different software version.

Pitch Correction Techniques

Here are a few techniques often used when correcting the pitch of a mix element. As always, don’t be afraid to experiment, since slight variations might fit better on a particular mix.

- Sometimes raising the formants (the harmonic resonance) of a voice can make the vocal sound a bit more exciting or breathier. It doesn’t actually change the pitch, just the placement of the voice’s harmonics.
- If the vocal is off enough that the pitch correction sounds robotic, try this trick from the old Harmonizer days. Copy the vocal to two additional tracks and spread them left and right. Tune one of the vocals up by 2 to 8 cents; tune the other down by 2 to 8 cents. This will smooth out an out-of-tune vocal and make it sound a lot thicker at the same time.
- Tuned vocals almost always sound better with a least a touch of reverb or delay on them.

Sound Replacement

Although you may be great at recording drums and have a great sounding studio with an excellent signal chain, the two chief variables in the recording are the drummer and his drums. No amount of technique or gear can overcome a bad sounding kit or a drummer that hits inconsistently, hence the importance of sound replacement software and a good sample library.

Drum replacement is a technique pioneered by the late great engineer Roger Nichols during the ’70s while working on Steely Dan records. In the days before DAWs, Roger developed his own hardware device called the Wendel that was an ultra-fast set of triggers that could replace real drum sounds with more consistent and better sounding ones from his library.

Today there are numerous drum replacement plugins that do the job equally as well, but it seems like every mixer has his own particular preference, be it Drumagog, Steven Slate’s Trigger, the Massey DRT, or any of the others that may be included as part of a DAW software package. The drum sound libraries can range from those that come with the software to personally made libraries consisting of the engineer’s own samples.
While the original idea was to totally replace a subpar drum track, most mixers now only do that as a last resort, opting instead to double the original sound with a sample to help keep the human feel and the basic drum sound intact. The new sampled drum acts as an acoustic EQ, changing the sound by augmentation instead of electronically changing the frequency bandwidth of the original recorded drum. Although some mixers try to delete the leakage between all of the drums of the kit, that’s what gives the kit its particular sound, and doubling the original kit with a sample helps to maintain some of that relationship.

Keeping The Sound Natural

It's very easy to make a drum track seem more like a drum machine than a human if you're not careful. For drum replacement to sound natural, the following must occur:

- The sample should sound better than the drum it's replacing.
- The sample should blend seamlessly with the other drums in the kit.
- No one should be able to tell that you've replaced the sound.

To replace drums, it's important to understand that any drum sound is composed of two different parts: the initial transient hit, and the resonant sustain and decay that make up the body of the sound. It's much easier to trigger a replacement drum if the initial transient hit is strong, since it will be clear to the software exactly when to trigger the sample.

> ***TIP**: Sometimes it's easier and more natural sounding to replace a sound manually by copying and pasting a hit or fill from another place in the song, especially if there are only a few hits that need replacing.*

It's usually best to record the replacement samples each to a new track rather than to keep the drum replacement plugin inserted as a plugin during the mix. This saves system resources (some sound replacement plugs are resource hogs) and makes your mix more transportable because the sounds are printed.

Sound Replacement Techniques

There are many techniques when it comes to sound replacement or enhancement. While most refer to drums, it's possible to use some of the same techniques in other situations as well if you're willing to experiment.

- If the original drum sounds okay soloed but disappears in the track, select a sample with a high initial transient. If the original tone of the drum doesn't fit the song, select a sample based on the character of the body of the sound after the initial hit.
- The simpler the drum pattern, the easier it is for the sound replacement plugin to recognize the parts.
- Most drums become brighter the harder they're hit; therefore, choose a sample that has these characteristics in order for it to sound natural.
- If the drummer plays the snare using mostly a sidestick, be sure that the sample has mostly sidestick as well. Timing is extremely important when adding a sample to the original sound, so take care to ensure that both transients line up together. Many sound replacement plugins have some latency, so be sure to check this closely and move the sample as necessary.
- The easiest way to find the transient is to use the Tab to Transient feature in Pro Tools, Hit Detection in Nuendo, or any other transient detection feature that your DAW has to offer. As an alternative, you can use any available Strip Silence–like feature on a copy of the original track to make the transients more distinct.
- Well-played drums with a few ghost notes will beat robotic-sounding, replaced drums any day.
- It's critical that you check the phase alignment of the sample against the original drum. Even though it may look to be in phase, your ears may tell you something completely different when the Phase button is selected. Snare drum is the hardest to augment/replace because of the nuances in how hard and how frequently the drummer hits it. Be sure to use a multi-sampled snare drum with multiple level variations to keep it sounding like it was played by a real drummer.
- If you don't have a multi-sampled snare or the sound doesn't have enough sample levels, you can make your own by duplicating the sample a few times and raising and lowering each one by 2 cents. Slightly changing the pitch gives the impression that the snare is hit a bit louder or softer.
- If the drummer plays a lot of ghost notes on the snare, sometimes it's best to make a copy of them and concentrate on replacing only those. This allows you to fine-tune the

sound replacement software only for ghost notes.

- Many times the best way to trigger the snare drum is by using the sound from the under-snare mic if one has been recorded. It's usually a highly transient signal that can be easily recognized by the sound replacement plugin.
- Sometimes using a gate on the original drum track can maintain just the transient hit, making the sample easier to trigger. This works especially well if you're trying to replace the body sound of the drum rather than the hit part.
- Sometimes compression of the drum track will provide a more consistent drum trigger.
- Many engineers always have the drummer give them solo hits of each drum while recording so they can use them as samples later.

***TIP**: You can make your own samples especially for the song by taking the best sounding hits from the recording of the kit you're mixing and triggering them as needed.*

Automation

Automation is the process of recording parameter adjustments so that your DAW software can automatically execute them during playback. Before automation, any mixing moves during a complex song had to be made by the engineer, the assistant, the band members, and anyone else who got their arms near the console. While this might have been a very “organic” way of mixing, it was not repeatable in the slightest, so engineers everywhere longed for a way that at least their fader moves could be memorized and played back later.

The first primitive automation on consoles came during the 1970s and was known as VCA automation, which revolved around a component called a voltage-controlled amplifier. It used two tracks of the tape machine to store the automation data and any updates, and while it remembered your fader moves, the faders themselves didn’t actually move. This was obviously confusing and became really unwieldy as consoles became larger and larger, plus having the VCAs in the signal path degraded the sound as well.

The next generation of console automation system actually provided moving faders (the first one was introduced by Neve and called NECAM), which were extremely expensive, as was the corresponding computer that memorized their movements. Don’t forget, this was in the pioneering days of computer science, when even 500 kilobytes of RAM (yes, kilobytes, not the far larger gigabytes that we’re used to today) cost thousands of dollars.

Modern mixing has always been a dynamic operation where a long list of console parameters are set during a mix, so it’s important to remember exactly each parameter’s setting position in order to make a remix of a song easier at a later date.

The first generation of console automation to make that possible was SSL’s Total Recall in 1981, which took a snapshot of all the switches and parameter controls on the desk. The problem was that the reset had to be done manually, usually by an assistant, which could take two or three hours to complete on a console with 48 or more channels.

For a brief period, resettable consoles became all the rage, where most of the console parameters would reset themselves automatically without the help of a human. Although this evolution came about in the analog days, it became much easier and cheaper to implement as digital came of age and is pretty much standard in most digital consoles today, even those costing as little as $1,000 or so.

But now we’re in the age of the DAW, where dynamic automation of virtually every parameter is the norm. This allows mixes to be more intricate than ever and take more time than ever as a result, but the pinpoint accuracy of every parameter movement during every millisecond of a mix is assured.

Fader Automation

Most automation systems, regardless of whether they're in hardware on a console or in a DAW, have the following modes:

- **Write**. Used the first time the automation data of a channel is recorded or when writing over the existing automation data.
- **Touch**. Writes automation data only while a fader is touched. The fader then returns to its previously automated position after it's been released.
- **Latch**. Starts writing automation data when a fader is touched and stays in that position after it's released.
- **Read**. The written automation moves are replayed.
- **Off**. The automation is temporarily disabled.

Besides the fader moves, all of the above modes include the fader and the mute button at the very least, but they extend to other parameters, such as panning, aux send levels, insert in/out, and other parameters, as well on most DAWs.

Generally speaking, most mixes are begun with the automation off until the basic balance is complete and any compression or effects are inserted. The first automated pass will be in the Write mode to initially record the static balance and parameter settings. From that point, most automation is written in Touch mode, where the changes are written as long as the parameter is being changed but then return to their initial state after the changes are completed. Occasionally a track may be written in Latch mode, where a brief change is required but it must then carry on until the end of the track.

It's not uncommon for a track to have many automation moves but require that the level of the entire track, including all the moves, be either raised or lowered in level. Most automation systems have what's known as a Trim control that allows the level of the entire track, or sometimes just a portion of it, to be changed by a selected amount (see Figure 11.16).

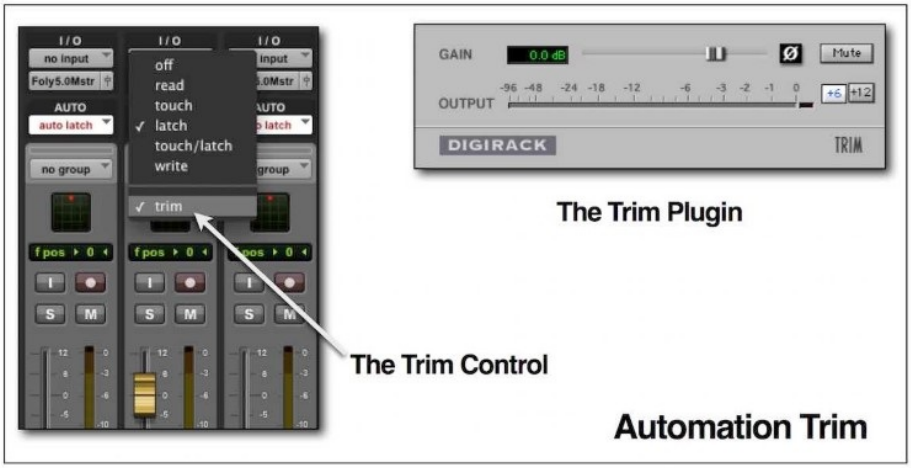

Figure 11.16: The automation Trim control
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Drawing The Automation

In many cases, it's much easier to draw the automation in than to use a fader if you need a high level of precision. This can work if you know exactly the curve you want the automation to look like. Sometimes it's easier to adjust the curve of what was previously recorded as fader automation, perhaps to trim the move shorter or longer (see Figure 11.17).

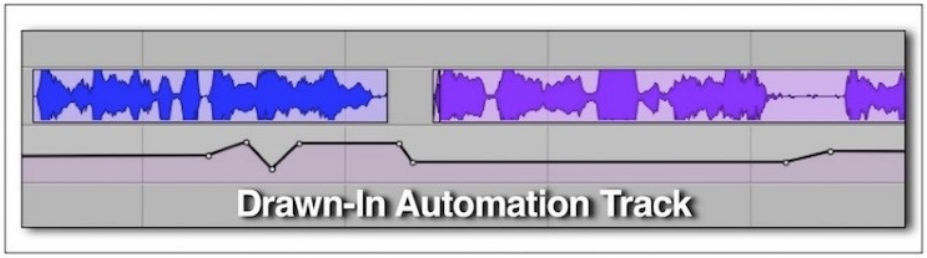

Figure 11.17: Drawing the automation
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Using Automation To Add Dynamics

For the most part, a mix where the faders remain more or less static can be boring and unexciting. Even before automation, mixers were constantly riding instrument and vocal faders during a mix to make sure they stood out in certain places or added an extra intensity to the mix. The best part about automation is that those moves can be exactly replicated on every playback.

> ***TIP**: The key to understanding how to use automation to add dynamics is by observing a performance by a great band. This will help you be able to hear all the nuances that the dynamics of the mix need for it to be exciting.*

Among the ways to add dynamics to a mix are:

- Slightly boost the rhythm section during fills, turnarounds, and even choruses (usually only a couple of dB is all that's necessary, but it depends upon the track).
- Boost the snare and toms during fills.
- Boost the kick, snare, or cymbals on accents or the downbeat of a new section.
- Duck the rhythm instruments during an instrument solo to help clear out space in the mix.
- Boost the hi-hat in parts where it's being struck and decrease it where it's not.
- Add additional reverb or delay to an instrument when it gets masked as other instruments are added to the mix.
- Pump a strumming rhythm guitar in time with the music, pushing it especially on 2 and 4, or push it on the upbeats (one AND two AND three AND…).
- Gently boost the fills or other instruments between vocal phrases. w Pull back the downbeat of a chorus if the drummer hits it too hard. w Pump the "4 AND" on a percussion track.

Automation Techniques

Each technique may apply to applications other than the one specified, so keep an open mind, especially when boosting the dynamics.

- Many old-school engineers use fader automation to control the dynamics of a vocal rather than use a compressor. This is accomplished by both riding the level of each phrase (or word or syllable) and riding the end of vocal phrases.
- Ride the bass to even out the energy differences of certain notes.
- Ride an instrument's sustain at the end of solos.
- Ride the reverb returns during parts where it gets lost.
- Use the Trim tool to adjust the automation of a track up or down slightly if necessary. No need to redo the track if the vibe is okay but the level isn't.
- Another way to adjust the automated track is to adjust the output volume of the last inserted plugin up or down as needed. If the track doesn't have any plugins, insert any one that has the ability to control its output without inserting the effect.

Gain Staging

Gain staging is the proper setting of the gain of each stage of the signal path so that no one section overloads. On an analog console, that would mean making sure that the input gain doesn't overload the preamp, so it wouldn't overload the equalizer section, so it wouldn't overload the panning amplifier, which wouldn't overload the fader buffer, which wouldn't overload the master buss. This is the reason why a pre-fader and after-fader listen (PFL and AFL) exist: to monitor different sections of the signal path to make sure there's no distortion.

Subgroups

Many engineers like to group certain instruments together because it's easier to control the level of one fader than many at the same time. Some engineers prefer groups (where the faders are all linked together), while others prefer subgroups (where the output of the certain individual channels are fed into a dedicated subgroup channel) or even combinations of both.

While all of these stages were slightly tweakable, one rule exists in the analog world that aptly applies to the digital as well.

The level of the channel faders should always stay below the subgroup or master fader.

This means that the level of the master fader should always be placed higher than each of the channel faders (see Figures 11.18, 11.19, and 11.20). While it might be okay if one or two channels is slightly above the master (it's almost inevitable in every mix), just a single channel with big chunks of EQ (like +10 of a frequency band) or an insert with an effects plugin with a level that's maxed can destroy any semblance of a good-sounding mix.

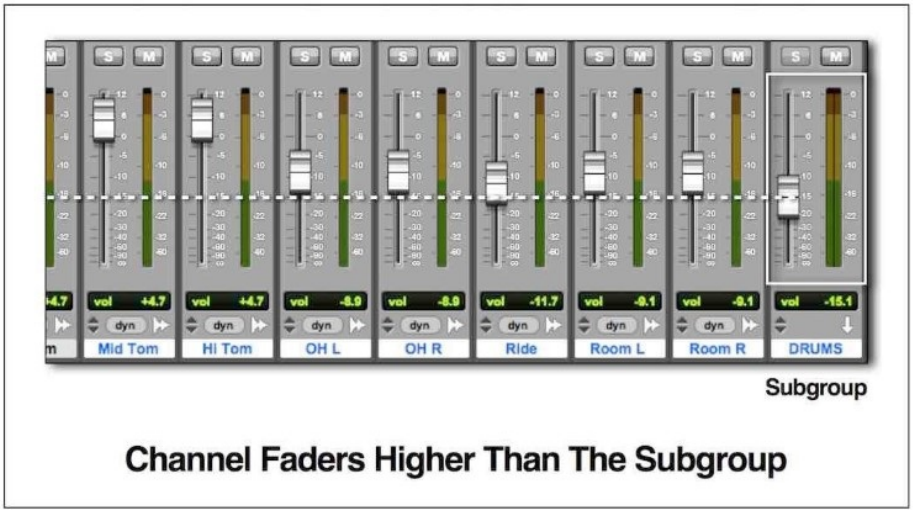

Figure 11.18: Channel faders too high, subgroup fader too low
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Figure 11.19: Subgroup too high, master fader too low
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Figure 11.20: Channel and subgroup faders at correct levels
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Because many large analog consoles have sufficient headroom these days, this rule hasn't always been religiously followed, but it has been a golden rule since day one of modern consoles. Not following it is the main reason why many mixes, especially those done in the box, loose fidelity. The master buss is overloaded!

Headroom

Speaking of headroom, that brings us to a second rule:

Leave lots of headroom.

Recording engineers in the digital world have been taught to increase the level of anything recorded or mixed to as close to 0dB full scale as possible in an effort to "use up all the bits." While this might have been useful back in the days of 16-bit recording, it doesn't apply as much today in the 24-bit and beyond world. The short reason for this is that if a piece wasn't recorded hot enough in the 16-bit days, it would be noisy as a result. That's not so much true in our 24-bit world. Today we can record at a level a lot less than 0dBFS and not have to worry about noise, and there's a great advantage in doing so: It gives us lots of headroom and therefore less distortion.

Headroom is our friend. We need it to preserve the super-fast transients that make up the first part of the sound of just about any instrument (but especially instruments such as drums and percussion). These transients can typically range as high as 20dB above what a VU meter might be telling you (peak meters are much closer to the true level), and recording too hot means cutting them off by going into overload if only for a millisecond or two. This results in not only a slightly dull recording but one that sounds less realistic as well. The solution: *more headroom.*

Headroom means that our average level might be –10dB or less on the meter, leaving plenty of room for transients above that. This concept is actually a holdover once again from the analog days, where a really good console might have a clipping point of +28dB at the master mix buss. Since 0dB VU = +4dB (trust me, it does), that means that you had a full 24dB of headroom before you ran into distortion as long as you kept your mix hovering around 0VU.

While leaving 24dB for headroom might be excessive in the digital world, leaving 10 or 15dB may not be. Since it's easy enough to make up the gain later, you won't increase the noise, and your mix will be cleaner, so why not try it?

> ***TIP**: When using large amounts of EQ or a plugin with a lot of gain, lower the channel fader or the plugin output rather than bringing up the others around it.*

Chapter 12

The Master Mix

Gone are the days of manual mixing, where the hands of not only the engineer but also the producer and all of the band members manned a fader or a mute button or a pan control to get the perfect mix. Gone are the days of massive numbers of takes of your mix to get your "keeper." Thanks to the digital workstation and the advanced state of console automation, if you work in a studio, the mix is (or at least should be) perfect before it ever gets committed to hard drive, analog tape, solid-state memory, or any other format yet to be devised.

That said, how do you know when you've reached that point of mix perfection—and when the mix has reached that point, what do you do with it then? That's exactly what this chapter will explore.

Eight Indicators That Your Mix Is Finished

One of the tougher things to decide when you're mixing is when the mix is finished. If you have a deadline, the decision is made for you as the clock ticks down, but if you have unlimited time or a deep-pocket budget, a mix can drag on forever.

Just when is a mix considered finished? Here are some guidelines:

1. **The groove of the song is solid.** The pulse of the song is strong and undeniable.
2. **You can distinctly hear every mix element.** Although some mix elements, such as pads, are sometimes meant to blend seamlessly into the track, most mix elements should be clearly heard.
3. **Every lyric and every note of every line or solo can be heard.** You don't want a single note buried. It all has to be crystal clear. Use your automation. That's what it was made for.
4. **The mix has punch.** The relationship between the bass and drums is in the right proportion and works well together to give the song a solid foundation.
5. **The mix has a focal point.** What's the most important element of the song? Make sure it's obvious to the listener.
6. **The mix has contrast.** If you have too much of the same effect on everything, the mix can sound washed out. Likewise, if your mix has the same intensity throughout, it can be boring to the listener. You need to have contrast between different elements, from dry to wet, from intense to less intense, to give the mix depth.
7. **All noises and glitches are eliminated.** This includes any count-offs, singer's breaths that seem out of place or predominant because of vocal compression, amp noise on guitar tracks before and after the guitar is playing, bad-sounding edits, and anything else that might take the listener's attention away from the track.
8. **You can play your mix against songs that you love, and it holds up.** This is perhaps the ultimate test. If you can get your mix in the same ballpark as many of your favorites (either things you've mixed or mixes from other artists) after you've passed the previous seven items, then you're probably home free.

How much time should all this take? In the end, most mixing pros figure at least a full day per song regardless of whether you're mixing in the box or mixing on an analog console, although it's still best to figure a day and a half per mix if you're mixing in a studio with an analog-style console and traditional hardware outboard gear. Of course, if you're mixing every session in your DAW as you go along during recording, then you might be finished before you know it, since all you may have to do is just tweak your mix a little to call it complete.

Competitive Level

As far back as the ’50s, mixers have strived to make their mixes hotter than their competitors’. That’s because if two songs are played back to back, the louder one is sometimes perceived to sound “better,” but the limitation of how loud a mix could actually be was determined by the delivery medium to the consumer. In the days of vinyl records, if a mix was too loud or had too much bass, the stylus would vibrate so much that the record would skip. When mixing too hot to analog tape, the sound would begin to softly distort and the high frequencies would begin to disappear. When digital audio and CDs came along, any attempt to mix beyond 0dB full scale would result in horrendously bad-sounding distortion.

Even with the built-in limitation of 0dB FS, over the years mixes have become hotter and hotter in perceived level, mostly because of a new digital technology that resulted in better and better limiters. Today’s digital “look ahead” limiters make it easy to set a maximum level (usually at 0.1 or 0.2dB FS) and never worry about digital overs or distortion again.

That being said, raising the competitive level (a mix level that’s as loud as your competitors’ mixes) used to be left to the mastering engineer. The mix engineer would hand off a mix that was deemed acceptable by all, and the level would then be raised to what was appropriate for the medium from there, regardless of whether the ultimate delivery medium to the consumer was a record, cassette, CD, or Internet stream. Part of the voodoo of the mastering engineer was his ability to make your mix louder than you, the mixer, were able to.

That doesn’t cut it these days. Artists, producers, and record execs want the mix to immediately sound not only “like a record,” but as loud as anything commercially released from the first rough mix onward. This is one of the reasons why the famous mix buss compressor on all SSL consoles became so popular. It was built like a typical mastering compressor to give your mix that “radio” sound as soon as you inserted it in the signal path.

Today, with the many powerful plugin compressor/limiters available, it’s all too easy to raise the level of your mix as loud as it will go, but just because you can do it doesn’t mean that you should.

Hypercompression

Too much buss compression or over-limiting either when mixing or mastering results in what has become known as hypercompression, where the waveform is cut off at the top or “flat-lined” (see Figure 12.1D). Hypercompression is to be avoided at all costs because:

- It can’t be undone later.
- It can suck the life out of a song, making it sound weaker instead of punchier.
- Many MP3 and online codecs have a hard time encoding hypercompressed material and insert unwanted side effects as a result.
- It leaves the mastering engineer no room to work, so his efforts are less effective.
- Many online music services now normalize the uploaded content, so being too loud has no advantage and can even sound worse than a song with lots of dynamics.

A hypercompressed track has little dynamic range, leaving it loud but lifeless and unexciting. On a DAW, it’s a constant waveform that fills up the DAW timeline region. Here’s how the levels have changed on recordings over the years (Figure 12.1).

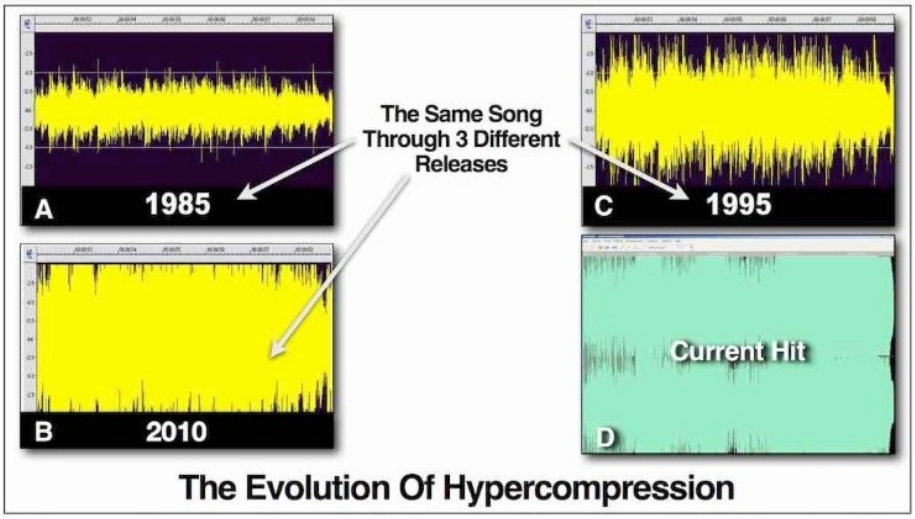

Figure 12.1: From very little compression to hypercompression
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Using LUFS For Mixing

You might have noticed that in the last few years the differences in level between television shows, commercials, and channels are pretty even, with no big jumps in volume between various programs. That's because viewers were complaining for years about the fact that there was a dramatic increase in level whenever a commercial aired because it was so compressed compared to the program that you were watching. Congress set out to do something about this, and in 2012 adopted a method to normalize those volume jumps that the European Broadcast Union put into place the year before – Loudness Unit Full Scale or LUFS.

LUFS (called LKFS in Europe) is a way to measure the perceived loudness of a program by measuring both the transient peaks and the steady-state program level over time using an a specially created algorithm. It's different from a normal meter in that it doesn't represent signal level – it measures how loud we perceive an audio program to be. For a broadcaster this is actually pretty serious, since if a station violates the mandated LUFS level of -23, it could possibly lose its broadcast license.

Even though LUFS was intended primarily for broadcast audio delivery, it has a new increased meaning in music production as well. Thanks to the fact that streaming services like Spotify and Apple Music are now normalizing the songs so the level is the same from tune to tune, there's no real benefit for compressing a song to within an inch of its life any more. In fact, less volume and more dynamic range are actually your friend.

Using a LUFS meter across you mix buss allows you to optimize your music mixes for a variety of platforms to be sure that they're always in the sweet spot for dynamic range. There are a number of LUFS meters on the market from TC Electronic, Waves or other developers, but one plugin in particular from Mastering The Mix called LEVELS (see Figure 12.2) shows not only a safe dynamic range for your mix, but also allows you to zero in on the level of the bass bands to make sure that the mix has just the right amount of low frequency energy.

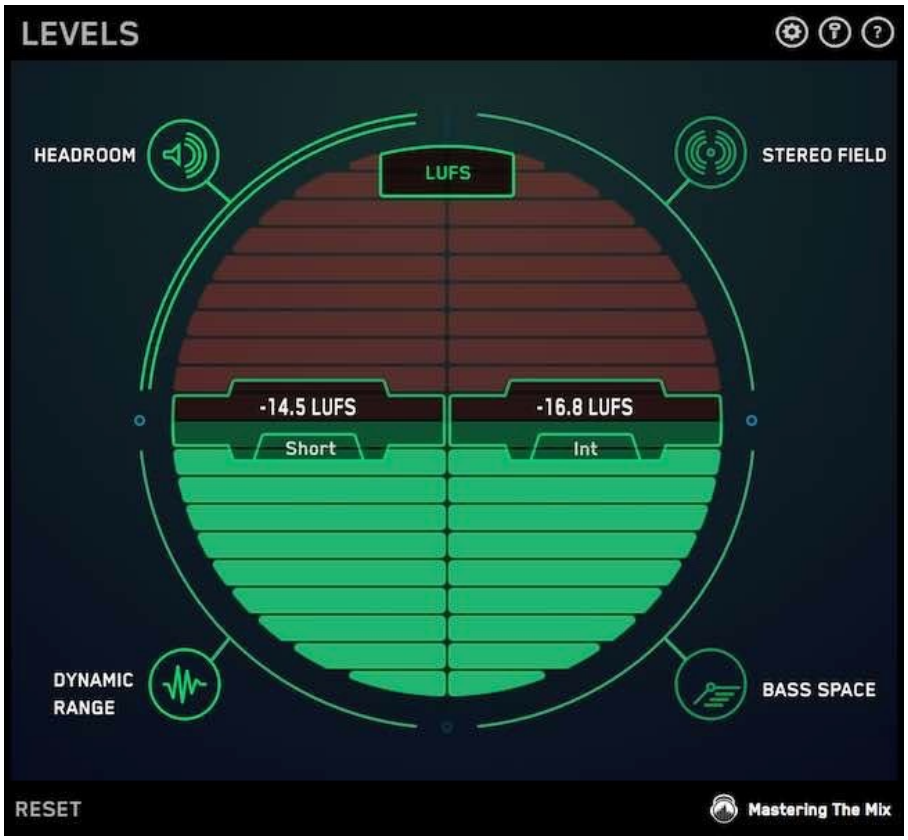

Figure 12.2: LUFS mixing with LEVELS plugin from Mastering The Mix
Courtesy Mastering The Mix

Tips For Hot Levels

That being said, it's important to attain relatively hot levels on mixes so clients can roughly compare them to existing records. Here are some tips used by mastering engineers to do just that:

- Set a compressor at a 1.5:1 or 2:1 ratio with a slow attack and release.
- Set the compressor for up to 5dB or less of compression.
- Raise the output of the compressor to the desired level.
- Feed the compressor into a limiter set to maximum of 0.1 or 0.2dBFS to prevent digital overs.
- Tweak the gain of the limiter to taste.
- DON'T HYPERCOMPRESS.

The real key is to use a compressor with a low compression ratio (1.5 or 2 to 1) and a slow attack and release time across the mix buss. This provides what's known as relative level, which means that it increases the perceived loudness of the track. The compressor then feeds a limiter (preferably a look-ahead digital one) set to clamp down at 0.1 or 0.2dBFS to prevent digital overs. Additional limiting is then used to taste (be careful—not too much!).

> ***TIP**: Make sure that this compressed "client mix" is clearly marked as such; then make another one with only your preferred buss compression without the extra level to send to mastering. This will give your client an idea of what the finished master will sound like, yet give the mastering engineer room to work so he can do his magic.*

Competitive Level Isn't What It Used To Be

Although everyone wants a hot mix, making it too hot might soon become a thing of the past in this online world that we now live. As stated before, many online music services like Apple Music and Spotify now normalize the uploaded content so it all plays at the same exactly the same level regardless of how hot or quiet the original uploaded master played. As a result, there's no longer a good reason to make the master levels extremely hot, and it fact, it's proven to actually be counterproductive. A song with more dynamic range and less level will actually end up sounding better than the louder one after the service's normalization process. Mastering engineers everywhere (as well as mixing engineers) will jump for joy as sanity returns to mixing levels and the "volume wars" finally end. We're not quite to that point yet, but for the first time in many years, it looks like some progress is being made.

Mastering

Technically speaking, mastering is the intermediate step between taking a mix from the studio and preparing it for replication, but it's really much more. Mastering is the process of turning a collection of songs into an album by making them sound like they belong together in tone, volume, and timing (spacing between songs in physical mediums such as CD and vinyl).

Mastering is not a plugin or a device that you run some music through and have it come out mastered. It's an art form that, when done conscientiously, relies mostly on the mastering engineer's skill, experience with various genres of music, and good taste.
In the early days of vinyl, mastering was a black art practiced by technical curmudgeons who mysteriously transferred music from the electronic medium of tape to the physical medium of vinyl. Just as mixing became more sophisticated, so did mastering.

Mastering engineers soon found ways to make the vinyl records louder by applying equalization and compression. Producers and artists began to take notice that certain records would actually sound louder than others on the radio, and if it played louder then it sounded better and maybe even sold better. With that, a new breed of mastering engineer was born, this one with some creative control and ability to influence the final sound of a record rather than being just a transfer jock from medium to medium.

Today's mastering engineer doesn't practice the black art of disc cutting much, but he's no less the wizard as he continues to shape and mold a project.

Why Do I Have to Master, Anyway?

Mastering is considered the final step in the creative process because it's the last chance to polish and fix a project. A principal function of mastering is making sure that the first impression opens the door for the artist, since most people audition music before they buy it. It's interesting to note that almost all of the major record labels and most of the larger indie labels still choose to master all of their projects with a major mastering house (see Figure 12.3), even though extensive mastering resources are widely available to just about any engineer.

Figure 12.3: Oasis Mastering, Burbank, CA.
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A project that has been mastered (especially at a top-flight mastering house) simply sounds better. It sounds complete, polished, and finished because the mastering engineer has added judicious amounts of EQ and compression to make your project bigger, fatter, richer, and louder. He has matched the levels of each song so every one on the album has the same apparent level. He has fixed the fades so that they're smooth. He has inserted the spreads (the time between each song) so that the songs now flow seamlessly together on a CD or playlist. He has sequenced the songs so they fall in the correct order, and he has made all the songs blend together into a cohesive unit. He has proofed your master before it goes to the replicator to make sure it's free of any glitches or noise, and he also has made and stored a backup in case anything should happen to your cherished master. All this happened so quickly and smoothly that you hardly knew it was happening.

Why can't you just use the "mastering" plugins on your workstation instead of going to a high-priced commercial mastering facility? Besides the above, there are many reasons why a real commercial mastering house produces a better product than home or self-mastering. First of all,

the mastering house is better equipped with super high quality D/A converters, ultra-smooth compressors and equalizer plugins (and even some hardware, too), and an exceptional monitoring system in a precision listening environment. In fact, the monitor system alone at these facilities may cost multiple times more than everything included in a typical home studio. Cost isn't the point here; quality is. You can rarely hear what you need to hear in most studios (even the biggest and best) to make the adjustments that you need at this point in the production chain.

Even though we often obsess about the gear, the mastering engineer is the real key to the process. This is all he does day in and day out. He has big ears because he masters for at least eight hours every day and knows his monitors the way you know your favorite pair of shoes. Also, his reference point of what constitutes a good sounding, balanced mix is finely honed thanks to working hours and hours on the best- and worst-sounding mixes of each genre of music.

While all of the above might seem like a way to discourage you from doing your own mastering, that's really not the case. In fact, the reference point of how the pros operate and why they're so successful is important to have. From there, you can determine whether you're better served by doing it yourself or using a pro.

If you attempt to master your own mix, use a different set of speakers from what you mixed on, because otherwise you'll only be compounding any frequency-response problems that the speakers might have.

Things To Remember Before Mastering

If you decide to use a mastering engineer, here are some tips to help you get the most out of your mastering session.

- **Don't over-EQ when mixing.** In general, mastering engineers can do a better job for you if your mix is on the dull side rather than too bright. Likewise, better to be light on the bottom end than to have too much.

- **Don't over-compress when mixing.** You might as well not even master if you've squashed it too much already. Hypercompression deprives the mastering engineer of one of his major abilities to help your project. Squash it for your friends and squash it for your clients, but leave some dynamics for your mastering engineer.

- **Come prepared.** Make sure all documentation and sequencing are complete before you get there. You'll make it easier on yourself and your mastering person if everything is well documented, and you'll save yourself some money, too. Be sure to include shipping instructions and record company identification numbers, and if your songs reside on hard disk as files, make sure that each file is properly ID'd for easy identification (especially if you're not there during the session).

- **Check your phase when mixing.** It can be a real shock when you get to the mastering studio, the engineer begins to check for mono compatibility, and the lead singer disappears from the mix because something is out of phase. Even though this was more of a problem in the days of vinyl and AM radio, it's still an important point because many so-called stereo sources, such as television, are either pseudo-stereo or only stereo some of the time. Check it and fix it if necessary before you get there.

- **Go to the mastering session if possible.** Most engineers and producers will go to the first few sessions when checking out a new mastering engineer to see whether he has the same musical and technical sensibilities. After that, a bond of trust will develop, and they will simply send the mix master with any instructions. That being said, you should go to all of the mastering sessions if possible, because it will always sound a bit different (and hopefully better) than what it sounded like during mix-down. Attending the session also allows for some final creative decisions that only you can make. (For example, "The kick is a little loud; see whether you can deemphasize it a bit," or "Let's squash the whole mix a little more to make this tune punchier.")

- **If you're making a vinyl record or CD, known the song sequence.** Sequencing (the order that the tunes appear on the CD or vinyl record) is especially important, and making this list beforehand will save you money in mastering time. Many engineer/producers have the mistaken impression that once the final mix is finished,

it's off to the mastering studio. There needs to be one additional session, known as the sequencing session, where you do any editing that's required (it's cheaper to do in your own DAW than during mastering) and listen to the various sequence possibilities. This is really important if you'll be releasing in multiple formats, such as CD and vinyl, or in different countries or territories, since they may require a different song order due to the two sides of the record.

- **Have your songs timed out.** If your project is going to be released on CD or vinyl, you want to make sure that your project can fit. Most CDs have a total time of just under 75 minutes, although it's possible to get an extended-time CD. (Be careful, you may have replication problems.) Obviously the available time decreases if you choose to include additional files on the ROM section of the disc. Cumulative time is important when mastering for vinyl because the mastering engineer must know the total time per side before he starts cutting due to the physical limitations of the disc. This is about 25 minutes per side if you want the record to be nice and loud.

- **Don't be afraid to ask for help.** If you're unsure of the amount of compression or EQ that you're applying while you're mixing, send the mastering engineer a mix of a song and ask for his opinion. He can guide you on what you got right and what you got wrong so that when you finally deliver the master to him, the mixes sound their best. Believe it or not, mastering engineers love to do this because it makes their job a lot easier in the long run.

Online Mastering

Like mixing, mastering costs vary a great deal. The costs for mastering with an A list mastering engineer can vary anywhere from about $350 an hour to about $500. As a result, you can expect the overall price for mastering an album at a top facility to come in anywhere from around $1,000 to around $5,000.

There now are some very adequate alternatives to spending a lot of dough on mastering however. By far the cheapest is to use one of the automated online mastering services like Landr, eMastering or Aria. The prices of these services vary anywhere from as low as $4 a song to a monthly fee of around $100, where you can upload an unlimited number. Depending upon the mix, the results can be surprisingly good, and you're usually allowed to try different mastering settings for just a single price so you're able to choose been a more aggressive or more dynamic mix.

Another alternative is to submit your finished tracks online for mastering at some of the top mastering houses like Abbey Road or Sterling Sound. Although you won't get one of the top engineers nor will be able to attend the session, you'll get a great job by a junior engineer at a price that's much reduced from the facility's normal rates.

Mixing For Internet Distribution

With CDs beginning their slow decline into history, it’s now important to have a handle on the online distribution options and how to mix for them. Downloads may soon follow CDs into delivery oblivion as streaming becomes the delivery method of choice, but the increased Internet bandwidth now makes it possible to deliver higher-quality audio than ever before, a trend that happily will continue. In the meantime, here are some tips on making master files for any Internet delivery method.

MP3 Encoding

Encoding an MP3 of your mix requires a bit of thought, some knowledge, and some experimentation. In the beginning days of online music, the idea was to encode the smallest file with the highest quality, which is, of course, the tricky part. That's changed a bit since broadband DSL and cable modems have become ubiquitous, but routinely 24-bit master files still need to be data compressed.

Here are some tips to get you started in the right direction so you won't have to try every possible parameter combination of the encoder. Remember, though, that the settings that might work great for one particular song or type of music might not work on another.

The Source File

Encoding for MP3 makes the quality of the master mix more of an issue because high-quality audio will be damaged much less by lossy encoding than low-quality audio will. Therefore, it's vitally important that you start with the best audio quality (meaning the highest sample rate and most bits) possible.

The Encode

It's also important to listen to your encode and perhaps even try a number of different parameter settings before settling on the final product. Compare it to the original source mix and make any additional changes you feel necessary. Sometimes a big, thick wall of sound encodes terribly, and you need to ease back on the compression of the source track. Other times, heavy compression can encode better than with a mix with more dynamics. There are a few predictions one can make after doing it for a while, but you can never be certain, so listening and adjusting is the only sure way.

Here are some things to consider if your mix is intended for MP3 distribution:

- Start with the highest quality audio file possible.
- Filter out the top end at whatever frequency works best (judge by ear). MP3 has the most difficulty with high frequencies, so filtering them out liberates lots of bits for encoding the lower and mid frequencies. You trade some top end for better quality in the rest of the spectrum.
- A really busy mix can lose punch after encoding. Sparse mixes, such as acoustic jazz trios, seem to retain more of the original audio oomph.
- Don't squash everything. Leave some dynamic range so the encoding algorithm has something to look at.

- Use multi-band compression or other dynamic spectral effects sparingly. They tend to confuse many encoding algorithms.
- Set your encoder for maximum quality, which allows it to process for best results. It doesn't take any longer, but the results can be worth it.

***TIP**: MP3 encoding almost always results in the post-encoded material being hotter than the original material. Limit the output of the mix intended for MP3 to 1dB, instead of the commonly used 0.1 or 0.2dB.*

Mastered For iTunes

Mastered for iTunes (many just call it “MFiT”) is a program in which the iTunes Store accepts high-resolution master files. Music files that are supplied at 96 kHz/24 bit will have a Mastered for iTunes icon (see Figure 12.4) placed beside them to identify them as such, although any sample rate that’s a 24-bit file will be considered.

Figure 12.4: The Mastered for iTunes icon

The reason why iTunes is asking for 96/24 files is so they can start with the highest-resolution source material for both a better encode and a bit of future proofing in the event that iTunes later converts to a different encode format or a higher encode resolution (it’s now 256 kb/s).

Mastered for iTunes doesn’t mean that the mixer, producer, or mastering facility does anything special to the master except to check what it will sound like before it’s submitted to iTunes and then check it later again before it’s posted in the iTunes Store.

All encoding for iTunes is still done by Apple, not by the mastering houses, record labels, or artists. The reason for this is to keep the encodes consistent and to prevent anyone from gaming the system by hacking the encoder, but also to avoid any potential legal problems that might occur when a mixer, producer, or mastering house sends the files directly to iTunes without the label’s permission or uses different submission specs.

As stated above, the mixer or mastering house doesn’t do any encoding directly, but Apple has provided a set of tools (go to apple.com/itunes/mastered-for-itunes) that can be used to hear what the final product will sound like when it’s encoded. That way, any adjustments can be made to the master before it’s submitted to iTunes to ensure a good encode.

So what are the tricks to get the best sound quality from an iTunes encode? It turns out that the

considerations are about the same as with MP3 encoding.

- **Turn it down a bit.** A song that's flat-lined at 0.1dB FS isn't going to encode as well as a song with some headroom. This is because the iTunes AAC encoder outputs a tad hotter than the source, so there's some inter-sample overloads that happen at that level that aren't detected on a typical peak meter, since all DACs respond differently to it. As a result, a level that doesn't trigger an over on your DAW's DAC may actually be an over on another playback unit. If you back it down to 0.5 or even 1dB, the encode will sound a lot better, and your listener probably won't be able to tell much of a difference in level anyway.

- **Don't squash the master too hard.** Masters with some dynamic range encode better. Masters that are squeezed to within an inch of their life don't; it's as simple as that. Listeners like it better, too.

- Although the new AAC encoder has a fantastic frequency response, sometimes **rolling off a little of the extreme top end** (16 kHz and above) can help the encode as well.

Mastered for iTunes is only an indication that a hi-res master was supplied; it's not a separate product. There will always be only one version of the song on iTunes, and it will be available at the same price regardless of whether it's "Mastered for iTunes." Mastered for iTunes doesn't mean you get to charge more or that iTunes charges you more. Everything is like it was before; you just supply a hi-res master so it sounds better.

Alternative Mixes

Until most engineers joined the DAW revolution, alternate mixes were a way of life for the mixer to cover himself in case a change in the mix was deemed necessary at a later time. Renting the studio and then setting up an analog console and all of the outboard gear just to raise the vocal 1dB in the bridge was both expensive and a time- consuming hassle. Plus, the setup was so inaccurate that the mix might not sound the same. All that changed with DAWs.

Now that so many mixers choose to mix at least partially in the box, mixes can be instantly recalled and fixes done in a matter of minutes. It's fast, cheap, and easy, and as a result, alternate mixes are no longer the norm except for the analog diehards. That said, it's important to understand about the most commonly used alternate mixes, because a producer or record label may still ask for them.

> *"There are just never enough options for some people. When you deal with major labels and managers, there's constantly this thing of, 'Can we get one with this in the second verse?', so I supply many levels of vocal up and vocal down."*
>
> *—Joe Chiccarelli*

Different Types Of Alternate Mixes

Alternate mixes were originally done to avoid having to redo the mix again at a later time because a change in the level of a mix element was requested. This means that any element that might later be questioned, such as a lead vocal, solo instrument, background vocals, or any other major part, was mixed with that part recorded slightly louder and again slightly softer. This is referred to as the up mix and the down mix. These increments are very small; 0.5dB to 1.5dB, but usually not much more.

With alternate mixes available, it's also possible to correct an otherwise perfect mix later by editing in a masked word or a chorus with louder background vocals. Many times an instrumental mix is used to splice out objectionable language, especially in the rap and hip-hop worlds.

Although many record companies ask for more or different versions, here's a version list for a mix that the label for a rock or country artist might ask for. Other types of music will have a similar version list that's appropriate for the genre.

- Album Version
- Album Version with Vocals Up
- Contemporary Hits Radio Mix (softer guitars)
- Album-Oriented Radio Mix (more guitars and more drums)
- Adult Contemporary Mix (minimum guitars, maximum keyboards and orchestration)
- TV Mix (minus the lead vocal)
- Vocals-Only Mix

The band, producer, or record label may also ask for additional versions, such as a pass without delays on the vocals in the chorus, more guitars in the vamp, or a version with the bass up. There is also a good chance that any singles will need a shortened radio edit as well.

Thanks to the virtues of the digital audio workstation and modern console automation, many engineers leave the up and down mixes to the assistants, since most of the hard work is already done.

> *"I still provide alternate mixes so I don't have to recall anything. Ninety-nine percent of the time recalls are not necessary anyway. The funny thing is that if you give them a vocal up, they'll use it. Give them a vocal even higher, and they'll use that. I still do it, but it's mainly for me*

for ‘just in case.’ I do not want to come back to remix. Once I’m done with a song, I’ve heard it so much that I don’t want to hear it ever again.”

—Benny Faccone

“I usually only give them the main mix, the TV mix, and the instrumental.”

—Jon Gass

“I used to deliver a vocal up a dB and a dB and a half, but it’s really not necessary anymore. I now deliver a lead vocal a cappella, a background a cappella, a TV track, and a main pass. Whoever is mastering the mix can put all those parts together to make a fix, as opposed to me delivering a lot more mixes.”

—Bob Brockman

Stems

In extreme cases, some mixers have resorted to the use of “stems” to keep everyone (mostly the record company) happy. A stems mix is a set of stereo submix files that usually consists of a stereo bed track and individual stereo tracks of the most important elements complete with effects. This allows for an easy remix later if it’s decided that the balance of the lead vocal or the solo is wrong.

Stems are widely used in film mixing because a music mixer usually can’t tell what may be too loud or masked by the dialog or sound-effects elements of the movie. The stems mix gives the dubbing mixer more control during the final mix if required. It’s typical for the dubbing stage to ask for a stereo (or even surround) rhythm submix, a submix of any lead instruments or voices with effects, the bass by itself, and any instruments with a lot of high frequencies, all isolated on their own submix.

> *“I still supply a vocal up and a vocal down, an a cappella, and lead a cappella, and instrumental, and then I’ll create stems for them so they can re-create any other combination for themselves later if they want.”*
>
> *—Jimmy Douglass*

PART II
THE INTERVIEWS

Chapter 13

Bob Brockman

"Bassy" Bob Brockman has a wide range of awards and credits, including more than 30 Grammy nominations with two wins, and an Oscar nomination. His many credits include Mary J. Blige, Toni Braxton, the Notorious B.I.G., Babyface, Aretha Franklin, Al Green, The O'Jays, Brian McKnight, Jodeci, Faith Hill, Korn, Laurie Anderson, Vanessa Williams, Christina Aguilera, Diddy, Herbie Hancock, the Fugees, Santana, and Sting. He's very much of the "old school/new school" in that his formative years as an engineer were spent in the analog world, but he's quite at home in the current digital one as well.

Can you hear the final product in your head when you begin to mix?

Yeah, probably. I think that I probably make some subconscious and non-verbal judgments when I first hear a song. I make a judgment on style and then go through a couple hours familiarizing myself with all the parts, then I try to see what's really crucial and what could be wallpaper. I then find whether there's something that's really important that I should make the listener aware of.

The first 20 years of my career I had a producer standing right next to me, telling me what parts were important. It's less so now because I see fewer people. I get sent digital files, and I sort of end up making those mix/production decisions on my own and end up delivering a more or less finished mix to the producer or the band; then I'll get notes on what to tweak.

Are you mixing on a console or in the box?

I can mix in the box if I have to, but it's certainly not my preferred way of mixing. What I'm into is a sort of hybrid mixing. I have a Neve 8816 [analog console] with 16 channels coming out of an Avid 192 D/A with an Alan Smart C2 compressor across the mix buss. Sixteen channels of analog makes a big difference to me in terms of power and depth of field. I still do mix quite a few things in the box, though, especially when I'm out traveling.

How much of the DAW do you use?

I'm very deep into the whole digital mixing process and do all of my work in Pro Tools. The plugins have stepped up a lot in the last few years, with the distortion and saturation plugs having improved immeasurably. They're now more transparent and not adding a lot of phase shift or distortion when you insert them. That was my problem with plugs before and why I would tend not to use them on phase-dependent things like drums and guitars. At a certain point, maybe the seventh or eighth hour of the mix, the whole thing would start to sound crunchy to me, so I would go in and bypass the plugs and realize that I was using them as a crutch to make things speak. Once you get things dialed in, by the end of the mix you don't need them as much, so there's a lot more sonic purity.

I often encourage young mixers to bypass their plugs and listen to what they have, especially in a

program like Logic, where when you open up a session it's already got three or four things inserted across every channel as a default.

There are certain things that you can't avoid, like de-essing. If you want to remove the sibilance from a vocal, you've got to use a de-esser, and the Massey de-esser is probably my favorite of all of them on the market. In fact, I'm a really big fan of Steven Massey's plugs. I use his L2007 [mastering limiter] a lot.

What are your go-to plugins?
I really like the Brainworx stuff, and I use a lot of the Waves stuff, like the API and the Jack Joseph Puig and Chris Lord-Alge Signature stuff. They don't always sound exactly like their analog counterparts, but that's not necessarily important. They definitely lend a power and tone change to what you put across them. I tend to use the CLA LA-3 plug more across the toms, snare, kick, guitars, and sometimes bass, because it has an interesting sound to it.

For reverbs I use the [Audio Ease] Altiverb a lot, as well as the stock Pro Tools Air Reverb, which I've made a lot of presets for over the years, so I have one for every situation. I also use the Lexicon vintage reverbs a lot.

Do you have certain effects that you always set up before you begin a mix?
I typically transfer all of my effects from one song to the next. I'll usually use an [SoundToys] EchoBoy or a [Massey] TD5 for delay. The [Waves] H-Delay and the [PSP] Lexicon PCM 42 are really nice as well. I usually have four or five delays, which vary from very tight to slap delays to timed things. I tweak the timing so it's either pushing or dragging a bit behind the beat. I usually have four or five reverbs all plugged in as well. I don't have any analog effects processing. It's all done in the box.

I do have a pair of Neve 1073s that I might insert across the stereo buss, but for the most part I'll just leave the equalization to the mastering guy. I try to get the EQ and the sound from what I'm doing to the individual tracks in the mix. I've never been much of a user of equalization over the years. I've worked with a lot of master buss equalizers like the Massenburg stuff, but there are so many equalization things that happen to the sound just by making adjustments on the Alan Smart [SSL-clone compressor]. It's such an amazing compressor with the way it grabs the low end and accentuates certain parts of the midrange or upper midrange, depending upon how fast or slow and the ratio.

I usually spend the last two hours of my mix not doing much mixing but listening and then making little adjustments to the master buss compressor and hearing what the impact is to all the parts. I definitely don't have a stock compression setting. I'm always moving the setting around on everything that I do. Each song has to have its own contour, I guess.

How hard are you hitting it?

That depends on the music. If I'm doing a dance record, I'm probably hitting it pretty hard. If I'm doing an aggressive rock record, then I'm sinking into it about 3 or 4 dB. If I'm doing something much more open or acoustic, I'm barely hitting it. Most of the effect is how it's putting the low-frequency information in check, which it does without the meter moving at all. Even when you're hitting it very lightly, it still has a dramatic effect on the music.

One thing I really like is The Glue [from Cytomic Sound Music Software], and I use it a lot. It's one plug that reminds me of the Alan Smart a little bit because it has a very dramatic effect. I also use the Steven Slate Master FG- X [Virtual Mastering Processor] plug, which is amazing. There was a time in the not-too-distant past when you would insert a plug across a channel and go "eh," because as you would open it up it sounded worse and worse. That has changed a lot in the last two or three years in plugin development.

Where do you start your mix from?

I've always mixed the whole thing together at the same time. In the old days I would set up everything across the console and very quickly sort of push things into place manually. I do the same thing now in Pro Tools, although I don't think it will ever be as intuitive a process in mixing in a DAW as on a console. There's something about sitting at a console with a bunch of faders and being able to grab something and move it, which is hard to replicate with a mouse, but I've gotten used to that workflow these days.

Another thing is I've been mixing with my eyes closed for most of my career. It was something that I started doing when I was 23 or 24 years old. I think I have an easier time visualizing the three-dimensional panorama that's coming out of the speakers. It's also a tool to help me to localize things. Somehow when I close my eyes it's easier for me to see an instrument or vocal by removing my eyes from the equation altogether.

Do you use a work surface as well?

I was using a couple of the Euphonix MC Mix controllers, but it got to the point where they were taking up a lot of space on the desk. I do a lot of work in the DAW graphically, especially in Pro Tools X, where I've gotten into Clip Gain. Automating vocals may have a thousand small moves to make sure the apparent volume of every single word is the same. The Clip Gain has a more natural sound to it, I think because you're adjusting the level of the vocal before it goes into the signal chain, whereas with the volume-based automation, you're adjusting it after it's already gone through the plugins. When you raise it up through Clip Gain, I think it behaves more like a console.

What monitors do you use?

I've been mixing on KRK E3s for 13 or 14 years. I use them with a 12-inch subwoofer that I set very lightly. With a lot of modern music there's a lot of sub information that's going on, and it helps if you can hear it a little more clearly.

How loud do you listen?
I listen pretty loud [laughs]. It's loud enough that I drive everybody out of the room for the first couple hours of the mix. I think it's important to know what the music sounds like loud, but once I've made the global decisions about the low end, then I'll usually go down pretty quiet, and that's where I tend to stay for the remainder of the mix. I might bring it back up when I'm getting closer to print just to make sure that there isn't anything that's too harsh on any of the really loud things of the mix.

What do you deliver to the client these days?
I used to deliver a vocal up a dB and a dB and a half, but it's really not necessary anymore. I now deliver a lead vocal a cappella, a background a cappella, a TV track, and a main pass. Whoever is mastering the mix can put all those parts together to make a fix, as opposed to me delivering a lot more mixes. I always thought it was absurd to deliver a mix with the lead vocal up a dB and half because it's usually not what you need. You usually only need a breath or a word or a chorus where it's just not loud enough. I don't think they ever got used anyway. I did that for about 15 years, but not anymore.

Has your philosophy changed from when you started to now, especially now with the digital workflow?
Yeah, it's changed a lot. There's so much detail in every mix now. I find myself automating almost every aspect of everything in the mix, whether it's changing the EQ of the backgrounds as they go into the chorus so they punch through, or automating the sends and returns to each of the effects. I'm also constantly filtering things like delay returns to make them brighter or duller at different sections of the song. It's now so much more involved and detailed than anything that I would've ever been able to accomplish on a normal analog console.

When I go back and listen to mixes that I did 15 years ago, I think, "Wow, that record is kind of fat," because it was done on an analog desk, but it's certainly not as involved and detailed as the mixes are now. It was enough to have a mix that had a great sound and a great feel to it 15 years ago, but everyone expects a lot more precision today. I also find that I'm doing 15 to 20 revisions now, and that just never happened earlier in my career. People just have the ability to endlessly make changes, and because they can, they do.

How long does it take you to do a mix these days?
It's an interesting question because it also begs the question, "What is a mix?" There was always a very clear delineation in the production world that I grew up in. When you'd do a mix, you wouldn't really do that much to it except maybe sample a snare drum to enhance it or something like that. The record was what it was.

Today there's a lot of cleanup stuff that's sort of expected as part of the mixer's job, which should be a production thing. Production has gotten a lot lazier in the past 5 or 10 years. Now it's unbelievable how much garbage, extraneous stuff, clicks and pops, and unlabeled tracks that you

get even in major projects. I don't know whether it's because you can have endless takes so you figure, "I'll just leave it to the mixer because he knows how to do that." Of course we do know how to do that, but it means the mix will take a lot more time.

If I'm mixing one of my productions where I've done the cleanup and organizational stuff ahead of time, I can usually mix it in six to eight hours. I find that with a lot of records that get delivered to me to mix, there are all these problems, like the drums aren't in sync or things have been thrown out of phase, and no one seems to have noticed. Maybe it's because there are too many tracks or maybe they just don't hear it. Other than the mastering engineer, I'm the last guy to see the record, so if those things are going to be dealt with, I have to deal with them, but it gets to be enormously time consuming.

How much do you replace sounds?

I guess it depends on the record and the style. If I have to replace something, I'll use the Slate Drum Tool, which is amazing. It has an incredible library, and it's really the only drum trigger that I've found that does it in time. If I get a rock record where the toms are less than inspiring, I'll replace them. If I get a snare that sounds like it was recorded in a closet, I'll add one that has a nice room sound to it.

More often than not these days, rock records are recorded in some austere conditions, like in a closet in their house. It's not a killer big room drum sound, and that's what you want because listeners have an expectation of how it should sound. You have to do whatever you have to do to get it to a place where people expect it be. It can be a really time-consuming process.

Chapter 14

Bob Bullock

Since he moved to Nashville in 1984, Bob Bullock has been one of the town's top engineers, trusted by the likes of Kenny Chesney, Shania Twain, George Strait, Reba McEntire, Hank Williams Jr., and Jimmy Buffett, among many others. Prior to that he saw a different side of the music world working in Los Angeles with acts such as the Tubes, Art Garfunkel, Seals and Crofts, Chick Corea, and REO Speedwagon. Now an instructor at several recording programs in Nashville, Bob continues to be a much in demand mixer. You can read more about Bob and see his complete discography on his website at bobbullock.net.

Can you hear the final mix in your head before you start?

Yes, but I don't necessarily know how I'm going to get there when I begin. I have to listen to the song and all its pieces before I have a vision for it, which is important because if I didn't have a vision of how I wanted it to sound, I'd just keep going around in circles.

Where do you start the mix from?

I start with the drums and bass and get the basic foundation, then add the guitars and piano or anything that would be considered part of the rhythm section. After that feels good, then I put a vocal in, because the style of music that I do is all vocal-driven, so the sooner I get it in the mix, the better. After that, I place any of the ear candy around the vocal and rhythm section.

I build the mix with balance, panning, EQ, and compression; then when I'm happy with all that, I'll start adding effects to give it depth. The final thing I do is add some rides on things like the vocal to build in a broader dynamic.

Do you start with the kick?

I found that it works best for me to keep the entire drum kit on while I'm adding EQ or compression. That doesn't mean that I won't solo the kick or snare, but what I don't do is just solo the kick and work on that for a while, then solo the snare and work on that. It usually starts with the whole kit so I can hear what the leakage is doing and hear what I need to do to make the kick or the snare pop through a little more. After I'm comfortable with the balance, then I'll solo the kick and maybe contour it a bit so it pops a little better, but I still try to make the drums sound like just one instrument.

How much do you do in the box these days?

I'm doing quite a lot in the box. In fact, I've got a couple of racks in the studio filled with outboard gear, but I'm in the process of letting a lot of it go. My new setup is mostly plugins based.

What are your go-to plugins?

I'm really a fan of the UA stuff. They've helped make an old-school analog guy like me embrace

digital. I definitely use their EMT 140 and 250 reverbs a lot, the LA-3As because they've always been my favorite guitar compressors, and occasionally the A800 tape emulator. I've got the Slate Digital mix buss plugins [the Virtual Console Collection] that I like a lot because it's not a compressor or an EQ, but it's bridging the gap between them. I don't use it every time, but when I feel that the mix may be lacking something, the Neve or Trident setting really helps. I like the Waves R-Vox and R-Bass, and I use a lot of the PSP stuff. One of my favorite delays is their PCM 42, and I like to use their EQs and compressors as well. All that said, if I can I try not to limit myself to just the go-to pieces and plugins that I usually use and try to change things up a bit.

What do you use for a controller?
Right now I'm using the Euphonix MC Control. In my old studio I was using some Mackie controllers that had 24 faders, but they were pretty slow because it was MIDI, and the MC controller is faster. I'll probably end up with a larger control surface so I can get more faders, though.

When you were working with Mutt [Lange] on Shania's albums, was it different from anything else you've done?
If I had never worked with Roger Nichols and Steely Dan as an assistant, I would probably say yes, because the biggest difference working with Mutt from most other sessions was the extreme attention to detail. Going back to when I used to work with Roger and Gary Katz and Walter Becker and Donald Fagen [of Steely Dan], the number of hours that used to go into everything was amazing, so to me working with Mutt wasn't all that unusual. For someone who's used to things moving faster, it would probably be torture. Mutt, for all his success, is basically a writer who produces what he writes or co-writes. His approach has always been not that different from Donald and Walter, or Paul Simon or Gino Vanelli or some people like that who put a lot of labor into their music.

We rarely get a chance to craft a record at those levels anymore because the budgets don't allow it. I'm just as much a fan of things that are done swiftly, but I admire those guys for the patience that it takes to get things perfect.

What's your approach to compression?
It depends on what I'm working on, but I tend to start with more compression and then pull back a good bit of it by the time I sign off on a mix. I always have a least some buss compression on all my mixes because it's expected of us now. When a client hears a mix, he wants it to pop. In the '70s we used to feel that we could control it better in mastering, and even though I add buss compression today, I make sure to leave a little for the mastering engineer to work with.

I always mix toward the buss compression knowing that it will be part of the dynamics, but on the individual tracks, it all depends on what I need to do to have it hit the buss compressor gently. I try to make them controlled and contoured enough so that when it hits the buss compressor there won't be anything extreme going on.

Are you aiming more for dynamic control or the coloring of sound?
For me it's for dynamic control. Back in the days of analog, there was something about mixing through something like a Neve 8068 to 1/4- or 1/2-inch tape that automatically gave it a fatness and dynamic control. In digital, because everything is so clean and clear, it means I have to use a little bit more individual compression to do the same thing. What I'm always trying to do is find ways to get a little of that analog fatness without using analog, and I have to say that using some of these mix buss plugins does that sometimes.

Do you use a lot of effects?
Yeah, I do, but it varies by the project and what the client expects. My personal taste is to use more layers, like using several reverbs to create one reverb sound, or using several short and long delays. My reverbs and effects usually end up coming from four to eight different sources. They'll be short, long, bright, dull, and everything you need to make an environment.

I use effects and reverb like we used to in that I'll set up an EMT 140 plate and have just about everything in the mix go to it a little bit. I guess if I was mixing a Nicki Minaj track I probably wouldn't do anything like that, but because everything I do is more organic, I find that feels the most natural.

Do you have an effects template?
No, I put them in as I go along, but what is common is that I might have a [Universal Audio EMT] 140 set to short and then a few others that might be longer. Same thing with delays. If I want to keep it hidden, I'll just use a shorter delay. If I want more depth, I'll have something short and long and blend them together. I learned a lot of that from [TEC Hall of Fame engineer] Roy Halee back in my assistant days because he used a lot of tape delays to give the mix depth.

Do you ever replace drums?
I'm a real fan of Drumagog to add samples to existing drums, but I rarely replace them. If someone sends me something where the drums are just unusable, then I'll have to, but generally I'll add a sample to add something or trigger an effect.

Do you monitor loudly or have any listening tricks?
When I'm starting off, I build my basic mix moderately loud. When I'm satisfied that everything is where I want it, then I listen pretty softly to see if something pokes out too much.

I'm pretty much a one system, near-field kind of guy now, since I began using the Carl Tatz Design Phantom Focus System (PFS) 4 based around a set of Dynaudio M1 Professional Monitors.

I also use a set of self-powered NHT M00s to listen as well. I either listen to them standing in front of them or even listen from another room. The PFS is definitely the main system, but occasionally I'll listen to the Moos to give me a closer-to-small system comparison to check

balances.

As far as listening tricks go, when I have all the songs of an album together, I'll make a CD and drive around in my car listening to it, keeping mental notes of things that I want to change. I personally have tried to get away from listening to things in too many places because it can get a little confusing.

Since so many people are listening on their phones and laptops these days, it helps to have a little bit of a perspective of what that sounds like too, but if we can make something really pristine and hi-fi on the big system, for the most part it's going to translate to those systems as well.

How long does it take you to do a mix?
That can vary, but I got used to having one day per song, although the PFS monitors has sped that up some. In the days of analog I'd start a mix, and later in the day the client would come in and we'd go in circles a little bit and make changes. If we still weren't sure, we'd come back the next morning when we were fresh, tweak it again, then we'd set up for another one. Under those circumstances, my ideal mix schedule would be 12 days to mix if there were 10 songs to do.

I don't mix that way much anymore because I do so much in the box. Now I kind of dabble with all the songs I have to mix for a couple of days, and then I'll fine-tune them one by one. If I like what my real LA-2A is doing rather than one of the pluginss, I'll just print it so I don't have any recall to worry about. Same with any outboard reverbs I might use; I'll just print them to audio tracks. When I mix this way it takes about five to ten hours per song.

Do you have a philosophy about mixing?
The general philosophy that I'd like to pass down is that all these plugins are just tools, and what might work for one song may not work for the next. There's no one preset that's always going to work on everything. Even if there was one, you still have to be wise enough to know when to run with it.

When I worked with Al Schmitt [as an assistant]—and there's no question about the quality of his recordings—he used very little processing. He did it all with balance and the right reverb. With Roy Halee, I would spend all day setting up tape machines for tape delay, and he also made great records. Humberto Gatica, who I also used to assist, used a lot of outboard gear, and he also made incredible sounding records. What I learned was that it's okay if your method is different from someone else's because it doesn't matter how you get there. Take all this information in, but in the end, use your ears.

Chapter 15

Joe Chiccarelli

Even though he may not have quite as high a profile as many other big-time mixers, engineer/producer Joe Chiccarelli's list of projects is equally as notable as the best of the best. With credits like the White Stripes, Alanis Morissette, the Strokes, Jason Mraz, Tori Amos, Beck, U2, Elton John, Oingo Boingo, the Shins, Frank Zappa, the Killers, Brian Setzer, and many more, chances are you've heard Joe's work more times than you know. In this updated interview, the nine-time Grammy winner describes how analog and digital combine perfectly during his mixes.

How much are you doing in the box these days?

A small portion. I definitely mix in the box when the budget only allows that, but I still love breaking it out on an analog console because it sounds so much better with the stuff that I do. I'm doing rock bands and organic music, and that stuff needs the size and aggression you get from a console. I don't feel that I can get the same results in the box, although I do a lot of premixing there. If you have 20 tracks of background vocals, it's better to premix them to stems there first. Even if I'm mixing in the box, I always try to use some sort of summing network because that improves the sound a lot.

How has the way you work changed now that so much goes on inside the box?

The digital world makes life easier and more cumbersome at the same time. The possibilities are endless, the ability to share sessions for collaboration is phenomenal, plugin technology has really come a long way, and the software has vastly improved. Everyone appreciates the sonic character of analog, but they don't want to or can't take the time for it.

I'll still mix to analog tape when the budget allows since I love the sound, and I do that on probably 75 percent of the projects I work on because of the glue it adds. Unfortunately, people can't always afford the cost of tape in the budget, and tape is not as consistent as it used to be.

How long does it take you to mix a track?

It really depends on the material, the number of tracks, and the arrangement. I try to work fast because I find that the longer it takes, the more I get into a sort of myopic mindset and get bogged down with the little details. You miss the vibe and the big picture and just suck the soul out of it, so I like to put it to bed in eight hours or so. In three hours I want it to sound like a record with the basic sounds and feel. In six hours I should have all the balances and it should start to sound finished. After that, the artist will come in for a listen.

As a tool, Pro Tools speeds up the process for you as a mixer, but the state of the world is indecision. I get lots of songs to mix that are 100 or 120 tracks with guitars upon guitars, and they want me to sort it all out. I don't need someone to print three mics for every guitar that they record. I think that one should have a concept of what the record should sound like and just go for it.

That's the thing that does take a bit more time these days in that there are so many possibilities, and people don't make decisions now because they know they don't have to commit to something until the very end. In fact, many of them don't even commit at the end of the mix and wind up taking a Pro Tools system into mastering to make the final decision there.

Where do you start your mix from?
I have no system. I really work differently for every project and every different type of music. It's a matter of finding out what the center of the song is or what makes the song tick. Sometimes you build it around the rhythm section; sometimes you build it around the vocal.

Usually what I do is put up all the faders first and get a pretty flat balance and try to hear it like a song. Then I make determinations from there whether to touch up what I have or rip it down and start again from the bottom.

If you're mixing an album, do you vary the sound from song to song or keep it all in the same sonic ballpark?
The approach varies from song to song, but I try to keep the same kind of reverbs and treatment for the drums. I try to keep some level of consistency, but again, I'm also treating every song differently as well. I personally like records that take you to 10 or 12 different places.

Do you add effects as you mix?
I usually try to start out with a flat track and then find the tracks that are boring and add some personality to them.

Do you have a standard effects setup?
Once again, it depends on the music. On some projects I'm not using any reverbs at all, while on some projects I might be putting all my reverbs through SansAmps or some other kind of cheap stuff. Sometimes I may only use a reverb and a delay for a whole record, but there are projects where I might use a plate, a live chamber, and a couple of old-school digital reverbs and tons of delays and plugin delays, so it's really song driven.

Do you have an approach to EQ?
It's weird. I just use whatever it takes for the particular project. It depends on what's been recorded, how well it was recorded, and how much work it needs. Bob Clearmountain is the genius for knowing what to touch and what not to touch, and I think that's really the secret—what to fix and what to leave alone. I find that the more I mix, the less I actually EQ, but I'm not afraid to bring up a Pultec and whack it up to +10 if something needs it.

One thing that I use is a spectrum analyzer that I put across my stereo buss that lets me know when the bottom end is right or the S's are too sibilant. I know what a lot of records look like on the analyzer, so I can tell when the overall frequency balance is right or when it might have some

obvious little hole in it. It definitely helps me sort out the 30 to 50Hz portion of the low end before it gets messy.

What's your approach to panning?
What I do is once I have my sounds and everything is sitting pretty well, I'll move the pans around a tiny bit. If I have something panned at 3 o'clock, I'll inch it a tiny sliver from where I had it just because I've found it can make things clearer that way. When you start moving panning around, it's almost like EQing something. I find that if I nudge it, it might get out of the way of something or even glue it together.

How much are you compressing things?
Compression is like this drug that you can't get enough of. [Laughs] You squish things, and it feels great and it sounds exciting, but the next day you come back and say, "Oh God, it's too much."

I do a lot of parallel compression, where I'll buss my drums to another stereo compressor and blend that in just under the uncompressed signal. Sometimes what I'll do if everything sounds good, but the bass and kick drum aren't locked together or big enough to glue the record together, is take the kick and bass, buss them together to a separate compressor, squish that a fair amount, and then blend it back in. I'll add a little bottom end to that if the record still isn't big enough. This helps fit the bass and kick lower on the record and gets it out of the way of the vocal.

On the mix buss I use a number of different compressors; sometimes it's an Alan Smart, sometimes it's a Focusrite Red 3, and sometimes it's something else that seems to work for the song. If I want something that's soft and warm, I'll use a tube compressor. If I want something that will give me a little more pop and glue, I'll use the Smart. In all cases, it's never more than a dB or two, because it's just for gluing the mix together. I'm not pumping it up dramatically to make it radio-sounding or anything like that, because I don't make those kinds of processed pop records. Most of what I do has real singers and songwriters, and they're not layered to death, so I don't have to compete with the latest flavor-of-the-month sound or artist.

Do you use more delays than reverbs?
It depends on the project. If it's a slick pop thing, then I might use a lot of reverbs; but if it's a rock band, then I might only use one reverb and maybe half a dozen delays. I'll try to do different things, like put only one instrument in the reverb or put a reverb in mono and pan it in the same space as the instrument. I like the mono reverb thing because it doesn't wash out the track, especially if you EQ the return of the reverb so that it doesn't conflict frequency-wise with the instrument. I've done some fun stuff, such as compress the returns of the reverb so that they pump and breathe with the signal that's there. It gives the reverb a cool envelope that comes up after the dry signal and doesn't fight too much with it.

What are you using for monitors these days?
I've been using the Tannoy AMS 10A's for quite a while, and I usually use those in conjunction

with NS10s and Auratones. Every once in a while I'll go up on the big speakers if those are good. I might get my sounds at a pretty moderate to loud volume, but when I'm getting balances it's always really soft. I listen in mono an awful lot and find that's great for balances. You can easily tell whether an instrument or vocal is fighting something else.

Do you have any listening tricks that you like to use?
Yeah, I might walk out of the control room and listen to it right outside the door. Things tend to pop out that aren't as obvious in the control room.

Do you have any go-to plugins?
I would say that the only thing I rely on is the UA stuff, just because I think their emulations are the best out there, and their library has so many options that I can go through super fast to find the perfect sound. I use a lot of their reverb and delay plugins. I like to use the EMT 250 on the snare, maybe the EMT 140 or a Roland Space Echo on a vocal. The dbx 160 has always been one of my favorites on bass guitar, so that's nice to have as a plugin, and the Trident A-Range EQ has always been one of my favorites, especially on electric guitars. Another favorite is the Little Labs Voice of God Bass Resonance plugin, which I love for putting some subharmonic frequency on an instrument.

When you're mixing on an analog console, where do you do most of your automation?
Both on the console and in the box. I find that I do basic moves for dynamics on the console, but all my fine-tuning is done on Pro Tools, especially the vocals. Stuff that requires tiny little maneuvers, I'll do almost all of in the computer.

Having plugins automated is a godsend and so easy compared to what you have to go through on an analog console. I'll automate things like feedback time, release time, or pre-delay time of the plate from section to section, which would be really difficult to do in the analog world.

When you get something into mix, how much cleanup do you have to do?
Too much, usually. [Laughs] One issue is organization, because the track labeling is often really poor, and I find I'm spending hours of prep time before I can even get into a mix. I get lots of tracks that are just labeled "Audio 1," "Audio 2," or "Bob 1" or "Bob 2." I don't care that it's Bob singing. I need to know exactly if it's a high harmony in the chorus. Same with the guitars. I'll get some that will be labeled "Matt" and "Jim." Now I have to play them to find out that they're even guitars, first of all, and then figure out what role they play in the song. I find that I have to spend two or three hours before I can sort it all out, even before I begin to mix.

Do you do much replacing drums?
I would say I do augmentation rather than replacement. If I use samples at all, I'm only mixing them 20 or 30 percent of the original, just to give them more consistency and more help to cut through the mix better. If I do it's with the Slate Trigger, which I've found to be the most consistent of all those kinds of programs, although I find I still have to tweak it a bit.

Are you doing many alternate mixes?

Constantly. There are just never enough options for some people. When you deal with major labels and managers, there's constantly this thing of, "Can we get one with this in the second verse?", so I supply many levels of vocal up and vocal down. It's just so much indecision. Back in the day when the technology was limited, you knew what you wanted and you just went for it, and that was it. Jack White, bless his heart, sticks with eight tracks. That's the record. Done.

Chapter 16

Richard Chycki

Richard Chycki has worked with everyone from Rush to Dream Theater to Aerosmith to Mick Jagger and more. In fact, he has 26 gold and platinum records from Rush alone! While he's worked on projects for many mega-stars, the last Dream Theater album may have shattered a record for complexity. While a home project that hits 50 or 60 tracks may feel immense, it's also interesting to compare that to what a really large project can be these days. You can find out more about Richard at richardchycki.com.

How much mixing in the box do you do?

Quite a lot. If I'm doing a surround project, then I do that entirely in the box. If it's in stereo, then I bounce between the box and and an SSL Sigma that I use for a hybrid approach.

What's your approach to mixing in surround?

I discuss it with the client first thing in a project because you get a better result if you record some extra elements with that in mind. Any additional microphones can give you some additional space to make it more authentic sounding so you feel like you're in the room with the band. I don't use fold downs for the stereo mix, I'll do stereo separately so that the client is happy with the imaging, then I'll put it out to surround with all the extra mics. As a result, I end up mixing the album twice, more or less.

For surround for a live album I like to keep a very strong audience perspective, so the band is in front and there are reflections in the rear. I'll spend a lot of time just listening to the hall mics to see what's going on and what's bouncing into the rear mics. I'll try to get a good recreation of what the house sound mixer is doing, along with the audience, so it really feels like you're there.

From a studio point a view it's a different approach, especially with a band like Rush where there's a chance to experiment. They want to maintain the feel of the original mixes, so there's definitely a heritage factor there, but try to get it interesting as well. When I mixed "Limelight" everyone thought that Geddy came into the studio to add a vocal, but it was just a harmonizer tuned down a fifth that was on the original but way down in the mix. When I heard it in the multitrack I thought I'd put it right in the rear speakers so every would hear it, and they did [laughs]. They thought we were re-recording it and it was sacrilegious somehow. It was always there on the album, and if you pop on a good set of headphones you can hear it, but when you remove it from the density of the front of the mix, it's a new experience.

How do you approach the LFE?

For live I'll put a little bit of kick drum, a little bit of bass, maybe a touch of guitars for the 'chunk' of the palm mute, and then I'll use it especially for pyro. I'll get in recordings where the visual of it is so amazing with a bomb going off and all you'll hear is a pip. I'll push a bunch of that down there so it sounds like what if looks like.

Are you adding sound effects?
That's been recording dependent. There are times when I didn't have to, and there have been times where, for whatever reason there's no low end, I'll add something to give it some power, but it's only in the LFE so your eyes and ears match up.

How do you handle the audience because there's two schools of thought. One is that you add in at the beginning and it becomes part of the bed of the song, and the other is that you add it in at the end as an enhancement.
That changes on a case by case basis. I'm looking more at the room ambience than anything. If it's something that I'm recording I'll go out and put 10 room mics up. What we're looking for there is a really great ambient sound, so we don't much worry about the audience. That's the biggest thing for me to achieve. When I turn this up, is it a good ambience and does it support the direct signal from the stage? If I have a good room sound, then the audience is usually on my side. If it's a crappy sounding room then it's not going to be very useful for a good mix.

Do you have certain plugins that you always use?
I love all the of the McDSP plugins. I use a lot of the Sonnox plugins, a lot of Universal Audio, and a few Waves as well. The UA SSL plugins sound really great as do the plugins from SSL. I really enjoy that SSL console grab and bite, and what I try to do is recreate that character in the box.

What's your approach when you're mixing something that you didn't track? Do you listen to the rough mixes or just go in cold?
I find that speaking with the artist is more beneficial than listening to the rough mix. So I try to find out what they want because if there's a mix already in existence then they're not happy with it, and I don't want to listen to something that they're not happy with.

If they mention something about the rough in our preliminary conversation, then I'll refer to the mix to find out what it is, but I find that if I learn what the band is about and where they want to go with it, it's easier to go from there.

How long does it take you to do a mix?
If everything is clean, sometimes songs mix themselves. I've had some that were 4 hours and others that were 3 days. There are no rules.

The great thing about mixing in the box is you can do what you do and the vocalist will come in and ask for the chorus to come up a dB. you just open it up and 90 seconds later, you're printed, so there's not that physical recall that can be incredibly time consuming.

Now the other side of that is that you've got all the minutiae that comes with it, like "The drummer says that the 37th high hat hit has to come up." [laughs]. You can have a situation where instead of having one, maybe two recalls, all of the sudden you're into 12. The time that the mix is

signed off on is getting longer because of the ability to do last minute changes.

What do you do when you have a lot of tracks?
I end up doing some mixing down before we mix. I may take as long as a week to go through and reduce the tracks and stem down similar ones to stereo if the elements are similar. If there are 3 different flavors of piano in a song, I'll often bounce them down to a stereo pair, that sort of thing. You tame it one way or another. It's no different than a movie. You comb through them a few times and reduce it down until you get what you hear in the theater. Slowly but surely it will happen. On the last Dream Theater album (*The Astonishing*) the biggest song track-wise topped out at 577 tracks!

That's mind-boggling.
It's mind-boggling for musicians. It's funny because I talk to anybody involved in making records and they're impressed. Anybody that's involved with making movies just goes, "Ha, rookie. That's small." All the movie guys are going, "Hey, I guess you're ready to come over to the dark side and work on films, brother."

I should clarify that when I say its 577 tracks, I don't mean it's 577 different elements. There's quite a few mics involved to record an orchestra or choir, and David's [arranger David Campbell] been doing this a long time, so every recording had extra microphones set up for contingencies. For instance, there's extra spot miking in case you have to pull something up in the mix of the orchestra.

During the mixing phase I was using three Pro Tools rigs, all locked together with word clock and time code. We'd do submixes on all of the three rigs, then those submixes were feeding an SSL Duality console at Germano Studios in New York City.

Was there any problems syncing three machines up?
No, they actually played very nicely together once all of the initial work clock issues were out of the way. Everything must use high quality cables. If you use any bad cables it's woefully apparent with word clock. Most of that work was just getting word clock set up and then getting time code to everything so all of the machines knew where they were at. The lock up was so fast because it's not mechanical like the old days on tape.

Were most of your effects in the box or were they external?
For this project I actually brought a lot of the effects from my studio. Things like a Lexicon PCM96X and 300L, a Bricasti M7 are the core of the spatial effects. Not to say that we didn't use things in the box. We actually did that a lot on the last record, but we were trying to get a different texture this time.

Can you hear the difference?
The funny thing is, yes, I did hear the difference. They sit in the mix really well. In a sparse mix it

becomes less of an issue, but if the mix becomes really complex, you do hear it. You can have more elements coexist in the same soundfield rather than have them jumbled together. I wish I knew what it was because I would bottle it and sell it to everybody [laughs]. There is something there. Is it psychoacoustic? I don't know, but it's one of those things that felt right, so we went with it instead of trying to fight it and say, "We have to do this in the box."

Did you find that the mix went faster or slower because you were on a console rather than in the box?
Being completely in the box is great because you can stem out more at a time. I would leave my assistant to do multiple passes of stems, and that takes time. Because of the shear size of it, the record ended up being 700 hours of mixing in 51 days. So there were a lot of really long days.

How long would it take to mix a typical song?
The first few songs took longer because you want to find the ultimate method and you're still getting vocal sounds set up and routing guitars and things like that. After a while the positioning and the numbers on the console start to get hammered into place, so it gets faster.

Some of the more complex songs actually went faster than some of much simpler songs, which you wouldn't expect. It feels done when the vibe is correct, and this record in particular very story oriented, so the energy level had to match where it is in the story. We started by mixing a few core tracks, and then after that we determined that we had to start at the beginning so that the mixes matched where they were in the story. Sometimes we'd be a few hours in and decide that we had to stop and change direction. That would take a bit of time.

Considering that the arrangements were quite large, did you find yourself muting any elements during the mix? Many times you can hear it in your head and it works, but when you finally hear the mix and have to match it to a story you find you have to pull things out.
That's a really good question. The short answer is yes. The tracks we got from David Campbell used a standard Decca Tree for strings, orchestra and brass and choir. In one of the choirs there was one soprano that was "iron lung." It was unbelievable how loud she was. Great tone, but she would just tear it up. In the Decca it was so overbearing that we had to resort to some of the spot mics to even out the choir. We'd have to pull down some of the elements as we got closer and closer to her because she was just so loud in the room. The choir wasn't recorded discretely, it was recorded as a group, so we had to go through and fine-tune elements which sometimes involved turning some stuff off.

What was the most fun?
The most fun was that as it got put together you could see that it was going to work. The fun was watching it all come together. It was such a huge undertaking that it fights the current trend of making things small. The band went the other way and said, "We're going to make this big."

For the last record, someone asked John [Petrucci] what it was like to work with me and one of

the highest compliments was that he said I know my stuff from a technical point of view but he said how much fun we had. Having fun in the studio affects the creativity. You only have a finite amount of energy, and the more energy dedicated to negative things, the less you have to create. I try to carry that to all the projects I work on. But really, the fun aspect is that we get to make records for a living. We're really blessed.

Chapter 17

Lee DeCarlo

From his days as chief engineer at LA's famed Record Plant in the heady '70s, Lee DeCarlo has put his definitive stamp on hit records from Aerosmith to John Lennon's famous Double Fantasy to releases by Rancid, Zakk Wylde, Black Sabbath, and Kenny Loggins. If you ever wondered where those sounds came from, Lee takes us on a trip to find out.

Where do you start to build your mix from?

The bass and drums. I'll put the bass and the drums up, and I'll get a rough sound on the drums real quick. I don't like to take a long time to do anything. I like to have it up and going. I just get the bass and drums so they just start to pump to where you can actually hear them breathing in and out with the tempo of the song, and as soon as I arrive at that, then I start adding the other stuff in.

How would you do that?

What I'll do is put the drums and bass in a limiter and just crush the hell out of it. Then I'll play with the release and the attack times until I can actually make that limiter pump in time with the music, so when the drummer hits the snare, it sucks down and you get a good crest on it. When he lets go of the snare, the ambience of the bass and the drums suck and shoot back up again. You can actually hear a [breathing sound] going on that was never there before. It was really there; it's just that you're augmenting it by using that limiter.

Are you using individual limiters on each track, or is it just a pair of stereo limiters that you use?

It's usually a mono limiter, and it's usually something like an 1176 or a Summit or something like that. It's whatever is handy at that particular point, brought up the center.

Do you have a method for setting levels?

Yes, I do. I'll have the drums around -5 (VU) with the snare drum constantly on the backbeat of the tune. From there, I'll build everything around it.

A lot of people that really haven't been doing this that long think that what you do is just turn things up and add stuff on top of other stuff. So much of mixing is what you take away, either level-wise or frequency-wise. There are so many things that you have to eliminate in order to make it all sit together and work.

What's your approach to using EQ?

When I'm mixing I use a minimal amount, but when I'm recording I'm radical when I'm EQing. I do a lot on the recording side, but I'm just redefining what I'm doing on the mixing side.

Do you use gates much?

Sometimes. I may have a gate augmenting the snare, but it's in such a weird fashion. I always use

the original sound of the snare drum, but I may have a gate on it that's so fast and has such a quick release that it sounds like somebody snapping their finger. I usually mix that in very low with just a ton of EQ on it, or use it just to send to an echo so that the snare drum doesn't have a lot of hi-hat or other things involved with it when it goes to the chamber.

Don't you do something special to your delays?
I always have people coming to me and asking, "How did you make John Lennon sound like that? What is the mojo filter that he puts on his voice?" There is no mojo filter. It's just John Lennon with a U 87 and a 15-ips delay. That's 133 milliseconds, or however many beats there are in the tune. I always put delays in the tempos of the songs.

Are you timing it to the snare drum?
Usually I'll find out what tempo the song is in; then I'll set my echoes so they pump. When the click happens, you get the backbeat or you get a 16th or a 32nd or a triplet or any sort of different returns for your echoes.

Then you're delaying the reverbs as well?
No, I very seldom delay an echo chamber. A lot of guys do, but I don't. I much prefer to use the chamber just as it is, but I do use a lot of different reverbs. I use somewhere around four or five different reverbs on everything I do.

How many delays would you be using?
Probably three or four different delays; it all depends. I like little tricks with delays as well. I like to leave out the delay on maybe the last line of a phrase. Then everything has delay on it until the very last word of a sentence or during an important statement.

How long does it take you to do a mix?
It all depends. I can mix three songs in a day, or I can mix one song in a day. To be really comfortable, I like to take about a day to mix a song and then go away and come back and finish it the next day. If you can't do a song a day, then you've got problems with the recording or problems with the band or problems with yourself.

What level do you usually listen at?
I like it loud. As a matter of fact, I'll start louder and work my way down. I'm always up there, but it's not crushing. People don't come in and bleed from the ears, but I'm over 100 (dB SPL).
What's your approach to panning?
I have several different approaches. I like to pan stuff around a lot, but I like to have the effects in mono. I like having things wide, but I don't like to have just a guitar on the right and the piano on the left. I've never been a big fan of that.

What are you trying to accomplish with effects? Are you trying to make everything bigger or to push things back in the mix?

Bigger, wider, deeper. Everything has to be bigger, always. Now, a lot of times I'll do stuff with no effects on it whatsoever, but I don't particularly like it. With effects you make a point about your music. Effects are makeup. It's cosmetic surgery. I can take a very great song by a very great band and mix it with no effects on it at all, and it'll sound good; and I can take the same song and mix it with effects, and it'll sound fantastic!
That's what effects are for. It's just makeup.

You're going for bigness rather than for depth, or both?
I'm going for pump, always. The better the band, the easier the pump happens. Nothing happens if the band doesn't play it in the pocket to start with. There's not a thing I can do to fix it if that doesn't happen.

Everything has to breathe. Songs have a life, and you have to develop that life within the song. Every single piece of music in the world breathes if it's played properly. A song is about something, and the trick is to capture what it's about and make it live. That's why mixing's an art and not a technology.

Do you have a special approach to treating lead instruments?
Yeah, sure. Bass and drums are the heartbeat, just like a human body, but the face is what everybody sees. It's kind of like looking at a pretty girl. You see her face and her body, but what makes her run is what's inside, so the pretty girl puts makeup on and gets a boob job. In essence, I give singers and guitar players boob jobs.

Chapter 18

Jimmy Douglass

After learning at the knee of engineer/producer Tom Dowd during Atlantic Records' glory days, four-time Grammy winner Jimmy Douglass (affectionately known as "the Senator") has gone on to become one of the most sought-after engineer/mixers in R&B, hip-hop, and rock. One of the few engineers who can cross genres with total ease and credibility, Jimmy has done records for artists as varied as Otis Redding, the Rolling Stones, Foreigner, Hall & Oates, Roxy Music, Rob Thomas, Snoop Dogg, Jay-Z, the Roots, Ludacris, Justin Timberlake, Timbaland, and Missy Elliott. But having old-school roots doesn't get in the way of Jimmy working in the modern world, as you'll see. You can read more about Jimmy and his Magic Mix Room at jimmydouglass.com.

How have things changed between when you first started and now in terms of your approach to mixing?

The urgency factor has definitely disappeared. Now we don't use a lot of musicians in the stuff that I do, we use machines. Everything is totally replaceable. As a matter of fact, you can erase a part that somebody played, and they'll just replace the part and nobody seems to care anymore. Back in the day, it was a major deal to replace anything.

And the rough-mix thing is becoming the nemesis of all of us now. Record companies want change and yet they don't want change. They want it to sound like the rough, but they want it to sound different. Someone will hand in a rough to a record company after taking a lot of time to make it sound good; then they'll hand it to me to do what I do. When I do what I do, they'll say, "Oh, it doesn't sound like the rough," and I'll think, "How am I going to beat a rough that somebody worked on for a month in six or seven hours?" As a result, I've been starting to match the rough. I never used to listen to them because then I couldn't really do what I do. Now it's the opposite. If you don't get close to the rough, the mix will probably never be accepted.

How long does it take you to do a mix?

It's beginning to change a little bit, but I'm a basic 10- or 12-hour man. Back in the day I could mix four or five songs in a day, but I just don't know how to do that anymore. Then again, back then you recorded what you wanted to hear in the end. Now people want to imagine things they don't hear.

One of the big things is that we might only actually spend maybe 4 hours of the 12 mixing the record because there are so many visitors and interruptions. People think nothing of stopping your mix and taking the time to play a whole record for a friend. I was in the groove, now I'm not in the groove anymore, and it takes some time to get back into it. We used to listen to records to get ideas or to try to emulate, but you were always working the whole time you were there. Now we might end up staying until 5 or 6 a.m., when we could've been done at like 1 in the afternoon. [Laughs]

Also, there are so many people hanging around or coming around to listen. Back in the day, the only people hanging around in the studio were part of the band or had a really good reason to be there. Now there are people who aren't connected to the project who are giving their opinion and who aren't really qualified to give an opinion.

Is mixing rap different from R&B or rock?
The tracking is so generic and sequenced and simple that the tracks have no real harmonics or overtones. There's nothing that's different, so it's really kind of simple. A lot of times I'll even use a stereo mix that the producer gave me because they can't find the original session to break the individual parts out, so all you're really doing is just putting the vocal on top. You have to try to make something sound really special out of something that's not.

That said, I've been doing a lot of EDM [electronic dance music] lately, and I really like it a lot. It has me excited and learning new stuff. As long as that happens, I know I'll be doing a good job.

Since you do all sorts of music, from rock to R&B to EDM, is your approach the same or do you prepare differently depending on the project?
The one thing that I do is something I call "tuning my ears." I listen to a lot of stuff in that particular genre to get to know what the particular sound of the day is. You want to sound contemporary and current, but you can't know what that is unless you listen to the records that the audience is digging at the moment. I'm not saying to copy it, but I tune my ears to know what the parameters are, so I listen to the genre to go, "Let's see what's considered cool today."

With some old-school guys, they're still making the same kind of records; but I'm making young records, and they're being made totally different from the way we used to do it. All the things they're doing I identify with because I was there, but they don't exist anymore.

I approach mixing records like fashion. This week tweed might be in, so even if I'm giving you the best silk in the world, you're not going to be interested.

Speaking of which, how much do you mix in the box?
I'm living in the box because I'm getting mixes that everyone wants recalls on. [Laughs] That's all anybody really cares about these days. "Give me a recall. I want to change something right now" is the attitude. They don't care what it sounds like. I have that great big [Neve] VR down in Miami, but I'm not using it much anymore.

Do you mix with a mouse or use a controller?
I have an SSL AWS that I use when I'm mixing in the box. I like to grab faders. If I want to put my fingers on five vocal faders like I used to, it's the only way to do it. That's a dexterity and relationship that I have that people who mix purely in the box don't have. It's like playing an instrument in that it's just a feel thing. I'm sure it can be done in the box, but it doesn't feel the

same. There's an eye/ear relationship to the distance of a fader's throw that's important.

Where do you control your automation from? Do you use faders or draw it in?
I'm trying to do as much as I can in the box, but when it comes down to vocal rides and things like that, I still have a better feel for using the faders rather than drawing in words. That works, but it feels so cold and calculated to me. Vocals have always been the most important part of a record to me anyway, so the way I control them has to feel right.

Do you have any tricks to make things sound good in the box?
I use an outboard summing amp for the warmth factor. I also use inserts with actual analog gear, but I always print it. If I use some effects in the analog domain or use the EQs off the board, I'll print it right back into the box. Sometimes the plugins just don't do what I want, so now that's what I'll do so I can still do recalls later.

Do you have any go-to plugins?
I have a lot of plugs only because I collect them all. I have two theories about how to use them. On some days I'll go to specific ones because I've had them for a while and I'm used to them; then on other days I think, "What's the difference? A digital EQ is a digital EQ." I go between days of hearing the difference and then not.

Do you use a lot of effects?
I'm from the old school where if you recorded it right, there's nothing to really add. I don't mix like that now, of course, because people want more bells and whistles and stuff. I always approach it like, "How can I give you what you put in here without changing it?"

I'm still using some of my analog effects because they do sound different. They don't have that harsh, cold, bright digital sound, especially flangers. The whole thing about a flanger is that it's nice and warm because the top end is cut off, but all you get with digital is top end. The same thing with digital delays. And I still use tape machines for slap- back. That also acts as my buss compression, since I always mix to a tape machine.

Are you compressing things more than you used to?
Absolutely. Everyone seems to want it smashed in your face now. I'm not saying it's a great thing, but I'm just doing what people want to hear. Buss compression has never been my thing, so my innate mixes usually sound lower than everyone else's right off the shelf. I put an [Waves] L2 or an L3 on it just to make it sound as loud as everything else, but I take it off when it goes to mastering. That lets the mastering engineer do his thing and bring the level up to where it needs to be.

How much EQ do you use? Are you old-school about it?
I use a lot of EQ. I like to bring the sound to me. I also add EQ, I don't subtract it. I know most of the world doesn't do that, but that's how I learned. Since I didn't know anything, the only way I

could tell what an EQ did was to turn it to where it brought something to me. Most engineers try to get something out of the way so they can hear other things.

That's really cool, but I don't know how to do that. If I'm listening to a vocal and I don't hear it in my face enough, I bring it out to me.

Do you replace drums or just add to them?
I never really did drum replacement much. There are so many samples out there that I figure that the producer already got the sounds exactly as he wanted it. If I have to replace all the sounds, then what is the producer doing in the first place? That's one of the great things about [producer] Timbaland. He has the innate ability to put together sounds that work. He can hear what instrument goes with what, but so many producers have no ear for that at all. I always try to remain as close to the original product as I can, because the people that created it would've already changed it if that's what they wanted. Besides, everyone has access to all the same toys and samples that I have. Since people have been living with a rough mix for two or three months with those particular sounds, if you change something they'll freak out on you.

Do you have to do a lot of fixes when you get a mix?
I've never been a guy who's done much of that. If you listen to my records you'll find that there's crap all through them. You can't do it all, so what's most important: the vibe of the record or the clicks and pops? I've always been a vibe-of-the-record man, so on my records there may be some things that others might find unacceptable that just don't bother me or my clients.

I do have assistants who rename and reorder the files for me before I start so I don't have to look at a whole bunch of files whose names mean nothing and have to figure out what they are. One of the biggest parts of the new mixer's world is having that support system around you. Without that support system, you can't compete.

That said, with the way things have changed, it comes to me at least sounding pretty decent to begin with.

What monitors do you use?
I'm still using my NS-10s, although I also have set of KRKs and a set of JBLs LSR 4300s with a sub that's ridiculous.

How many versions of a mix do you do?
I still supply a vocal up and a vocal down, an a cappella, and lead a cappella, and an instrumental, and then I'll create stems for them so they [the client] can re-create any other combination for themselves later if they want.

The digital recording revolution is a great triumph for audio. It brings such great flexibility and versatility to the craft, but it also brings the element of mediocrity in the creativity part of the art.

No one has to make any decisions now, and consequently, nobody really knows or thinks about what they really want. It’s digital, they think, so you can always change it later.

Chapter 19

Benny Faccone

Engineer Benny Faccone is unique in that he's Canadian from Montreal, but most of what he works on is Latin. From Luis Miguel to Ricky Martin to the Latin superstar rock band Mana to Santana to the Spanish remixes for Boys II Men, Toni Braxton, and Sting, Benny's 10-time Grammy-winning work is heard far and wide around the Latin world. Find out more about Benny at bennyfaccone.com.

What's the difference between doing a song in Spanish and one in English?

First of all, the way they sing in Spanish is totally different than English. The syllables don't fit the same way. With English music, it feels like the voice fits right into the music rhythmically. That doesn't happen with Spanish singing because it has different accents with harder S's. You have to treat it a different way on the mixing side by building the rhythm track around it. It's a different flavor with a different kind of emotion.

Are there any other differences between doing an American record and a Latin one?

Everything I do is treated like an American record. The two have gotten a lot closer in terms of the music, but they've also become way different because Latin artists are so concerned with their vocal level. The vocals have gotten so loud on the Latin mixes that it borders on ridiculous sometimes.

Do you just do Latin pop or do you do any traditional salsa?

As a matter of fact, I do everything. The Latin field is not very specific like the American market, where you do one type of thing and that's all you do. In Latin music, you just do it all. I've even done a couple of mariachi records. There were a few records where they wanted some traditional salsa, and the only way to get it was to go to Puerto Rico and do it there. I had to get some ideas of how to do it from some engineers down there since they have very specific placement for a lot of the instruments.

And what is that exactly?

They've got two or three different ways of doing it, but the things that stay the same are that the shaker and the bongos are always in the middle.

Do you have a philosophy or an approach to mixing?

The only approach is to try to figure out the direction of the song, develop a groove, and build it like a house. It's almost like a musician who picks up a guitar and tries to play. He may have the chart in front of him, but soon he has to go beyond the notes in order to get creative. It's the same thing with mixing. It's not just a thing of setting levels anymore, but more about trying to get the energy of the song across. Anybody can make the bass and the drums even out.

How do you build your mix?

It really is like building a house. You've got to get the foundation of bass and drums and then whatever the most important part of the song is, like the vocalist, and you've got to build around that. I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom.

Sometimes you can have a really thin kick by itself, but when you put the bass with it, it now seems to have enough bottom because the bass has more bottom end. I build the drums on top of that. After I do the bass and drums, then I get the vocal up and then build everything from there. A lot of mixers just put the music up first, but as soon as you put the vocal up, the levels become totally different. After all the elements are in, I spend maybe a couple of hours just listening to the song like an average listener would, and I keep making improvements.

Do you have a method for setting levels?
Yeah, I have a starting point. I usually start with the bass at about –5 VU and the kick at about –5 [on an analog console]. The combination of the two, if it's right, should hit about –3 or so. By the time the whole song gets put together and I've used the automation to adjust levels, I've trimmed everything back somewhat. The bass could be hitting –7 if I solo it after it's all done.

Do you put the snare at about the same level as the kick?
No, that's more a question of feel than level, because it has so many transients that it could be reading –10 and still be too loud.

What's your approach to EQ? Do you have certain frequencies that you always come back to on certain instruments?
Yeah, as a starting point, but I'll do whatever it takes, depending on how it was recorded. For bass I use a combination of a low frequency, usually about 50Hz, with a limiter so it'll stay tight but still give it the big bottom. Add a little 7k if you want a bit of the string sound, and between 1.5k and 3k to give it some snap.

For the kick, I like to have bottom on that, too. I'll add a little at 100 and take some off at 400, depending on the sound. Sometimes I even take all the 400 out, which makes it very wide; then I'll add some point at 3k or 5K.

On the snare I give it some 10k on the top end for some snap and 125 Hz on the bottom to fill it out a little more.

For guitars, usually 1.5k gives it that present kind of sound. Pianos and keyboards vary so much that it all depends on how it feels in the track.

For vocals, it really depends if it's male or female. If they sing really low, I don't add as much bottom end. Usually I always take some off at about 20 Hz to get rid of any rumble, but anything on up, it really all depends on the singer. I might add a little bit in the 4k to 6k range as well.

How much has your approach to mixing changed since most of the recording world has switched to Pro Tools?
The major thing that's changed is the way I approach the bottom end. I never used to put a stereo limiter across the mix buss, and if I did, it was really light. Now you almost have to make it sound like a finished product and make it tighter sounding, so I don't spend as much time working on the bottom end as I once did.

I'm also concerned about how things translate to the radio more these days. Nobody's listening to CDs anymore, but they do listen in their car. I have to readjust how I do things because all the compression and EQ that radio uses is based on the 99 percent of people mixing in the box. When I hear my stuff on the radio, it's got more compression and it's fatter (mostly because I like it that way), so the station's compressor is hitting it harder, which means I have to readjust what I do so it works better on the radio.

What compressor are you using on the stereo buss?
I have an old SSL that I use 90 percent of the time. If I need something that's not as aggressive, I rent a Fairchild or something like that.

Do you begin your mix with the compressor across the buss?
No, after I have the foundation built, I put the buss compressor in. I hit it so it just barely moves the meter. With the SSL, the louder you crank the gain, the grainier the sound gets, so that's why I always try to start as loud as I can so I don't have to add more than 2 or 4 dB of level at the most.

What do you use in the box?
I use plugins for effects, but very few compressors or EQs. Only if I run out of outboard gear or stuff on the console will I resort to what's in the box. I do use more effects now because it's easier to do it in Pro Tools than hooking up the outboard gear. [Laughs] I like using flangers and Moogerfoogers and stuff like that to give it a little bit of something different. If I can, I try not to insert it on the same channel so there's some stereo space to it.

Are you mixing to only Pro Tools?
In my studio [The Cavern] I am. If I mix at Conway or one of the other studios that I like to work at, I'll mix to 1/2-inch [tape], too. Pro Tools is winning out more and more instead of 1/2-inch tape because of the clarity. On something I did recently we mastered one song for seven or eight hours, going back and forth between the 1/2-inch and the Pro Tools file. The 1/2-inch had a warmth to it, while the Pro Tools file had a clarity. It was thinner but it was clearer. Neither one was better, but they were both so different. We ended up trying to add the warmth of the 1/2-inch to the Pro Tools master, but after trying different things, it still didn't work.

What's your approach to individual track compression?
Limit the heck out of everything. [Laughs] I like to compress everything just to keep it smooth and

controlled, not to get rid of the dynamics.

Usually I use around a 4:1 ratio on pretty much everything I do. Sometimes on guitars I go to 8:1. On the kick and the snare I try not to hit it too hard because the snare really darkens up. It's more for control, to keep it consistent. On the bass, I hit it a little harder, just to push it up front a little more. Everything else is more for control rather than to have it stick right up in your face.

Do you have any special effects tricks that you use?
I use a lot of the old PCM 42s on guitars for a very short slap delay. It's mono, but it sounds really big. I use something like 4, 8, 11 milliseconds, so it doesn't sound like a delay. Sometimes I use as much as 28 milliseconds on a power guitar. You stereo it out, and it'll sound like two guitars on either side of the speakers.

Is there a certain listening level at which you always listen?
Yeah, I listen at a fairly modest level, not loud and not soft. When I start the mix, I crank it on the big speakers to kinda get hyped a little bit and check out the bottom end; then I'll slowly start listening softer and softer.

Does anything change for you when you get something to mix that you haven't recorded?
When I get a mix in, I keep some of the automation but take most of the effects off. It seems like nobody builds a song from the recording stage anymore. They throw a lot on, and you have to figure out what to do. That makes it really difficult because someone has been living with the song for months, listening to their version of the mix, then they give it to you and within a day you're expected to do whatever you want. It's almost like editing a book before reading it. What do I take out? If I take something out, how do I know if it's something that's needed somewhere else in the song? I almost don't know what to take out or put in until I finish the song, but you can't finish the song until you decide those things. It's a catch-22.

There's so much more wasted time now, because when they send you the session file, you have to prepare it. Since they're not in the room with you, after you send the mix out you have to wait for them to get back to you before you can do the tweaks. I have to give parameters because I'm in analog and I can't wait that long.

Do you do many alternate mixes?
I still provide alternate mixes so I don't have to recall anything. Ninety-nine percent of the time, recalls are not necessary anyway. The funny thing is that if you give them a vocal up, they'll use it. Give them a vocal even higher, and they'll use that. I still do it, but it's mainly for me for "just in case." I do not want to come back to remix. Once I'm done with a song, I've heard it so much that I don't want to hear it ever again.

Chapter 20

Jerry Finn

From his mixing debut on Green Day's *Dookie* to producing and mixing Rancid's *Out Come the Wolves* and *Life Won't Wait* to his work with the Presidents of the United States, the Goo Goo Dolls, Blink 182, the Offspring, and Beck, Jerry Finn's distinctive sound has been loved by artists and listeners alike. Unfortunately, Jerry passed away suddenly in 2008 just after finishing Morrissey's *Years of Refusal*. I wanted to keep his interview in the book even though it's not current, both as a tribute to his great work and because the interview has a lot of really useful info.

Do you usually have to work fast because of the budget?
Not usually. I generally take about 10 to 12 days to mix a record. Some take less; some take more. *Dookie* I think we did in nine days. *Insomniac* took 11 days.

I mixed Beck for a PBS show called *Sessions at West 54th*. We were supposed to only mix four songs in one day, and it went so well that we ended up mixing seven songs in ten hours, and it came out great. The stuff was recorded really well, and his band had actually just gotten off a year and a half tour, so they just nailed it so it didn't really require any fixing. And Beck is someone who really trusts his instincts so he doesn't sit there second-guessing himself. We just went straight for what sounded right and just nailed it.

Before you start a mix, can you hear the final product in your head?
Yeah, that's actually one of the requirements for me to feel comfortable going into a record. When I'm sent rough mixes, I really need to hear where I would take it in order to feel comfortable. Sometimes the band tells you what they want and the producer tells you what he wants and the A&R guy tells you what he wants, and they're all completely different things. That can be a bit frightening because you end up being the punching bag for their arguments. [Laughs] But I usually can hear the final mastered record from day one, and then it's just trying to get the sound that's in my head to come out of the speakers.

Where are you starting your mix from? Do you start from the kick drum, the overheads...?
Just out of habit, I probably start at the far left of the console with the kick and start working my way across. Lately, I've tried to put the vocal in early in order to create the mix more around that. In a lot of the punk-rock stuff you get the track slamming and then you just sort of drop the vocal on top. But for the more pop stuff, I've found that approach doesn't work as well because the vocal really needs to sell the song, so I've been trying to discipline myself to put the vocal up early on, before I even have the bass and guitars in and kind of then carve those around the vocals.

One thing that I do with drums, though, is try to get the room mics in early on before I start adding reverbs and stuff like that to the snare. Unfortunately, recording drums is sort of becoming a lost

art. As engineers have gotten more and more dependent on samples and loops and drum machines, and with more recording being done in home studios, the thing that always suffers is the drums.

Do you get a lot of stuff that's done in garages or homes?
Not so much, but I do get stuff where the band thought that going to a good studio would be all they needed, and they didn't really think about the engineer they hired, so I've seen some engineers that get in over their heads. I was actually a drummer myself when I played in bands so I tend to be real anal about the drum sounds.

After you put the drums up, where do you usually go from there?
I'll get the drums happening to where they have some ambience, then put the vocal up and get that to where that's sitting right. At that point I'll start with the bass and make sure that the kick and the bass are occupying their own territory and not fighting each other. Sometimes to my surprise I've nailed it and it all falls together, and then other times when I get the guitars in there, they eat up a lot of the ambience on the drums. Most of the bands I work with tend to have several tracks of very distorted guitars, and they want them all real loud, so then I have to go back to the drums and kind of adjust for that.

How do you deal with that when you get a lot of real big, crunchy guitars?
When every guy in the band thinks he's the loudest, that's when I know I've nailed the mix. I've always tried to just make it so that you don't have to fight to hear anything. On certain parts of the song maybe I will bury something a little bit or push something a little louder for tension to kinda pull you into the next part, but overall I try to make it so you can hear everything all the time, and that generally comes through EQ. I'll find the bite in the guitar and make sure that the snare isn't also occupying that same range, then I'll make sure the low end on the guitars doesn't muddy up where the bass is sitting. I also have to keep the kick and snare really punchy to kind of cut through all the wall of guitars by multing them off and hard-compressing and gating them and sneaking them back in under everything.

Do you find you use compression on a lot of things?
Yeah. I'm a big compressor fan. I think that the sound of modern records today is compression. Audio purists talk about how crunchy compression and EQ are, but if you listen to one of those jazz or blues records that are done by the audiophile labels, there's no way they could ever compete on modern radio even though they sound amazing. And unfortunately, all the phase shift and pumping and brightening that's imparted by EQ and compression is what modern records sound like. Every time I try to be a purist and go, "You know, I'm not gonna compress that," the band comes in and goes, "Why isn't that compressed?" So yeah, I compress the [stereo] buss, although I'm very sparing on certain records. *Dookie* for Green Day had no compression on the buss at all, and the Superdrag record that I produced and mixed [*Head Trip in Every Key*] didn't have any either, but if I think it's appropriate for the music, I'll get it on there.

Are you compressing everything else individually as well?

Lately what I've gotten into doing more is multing it off, like I said [for parallel compression]. The kick and snare I'll put through maybe a [dbx] 160 and very lightly compress it, maybe pulling down half to 1 dB; then I'll mult them off and go through a new [dbx] 160S and really compress those and sneak them up underneath so you're basically hearing the character of the drum you recorded rather than this bastardized version of it. I also send all of my dry drum tracks—not the rooms or overheads, but the kick, snare, and toms—through another compressor and sneak that in to give the kit an overall sound.

Distorted guitars I don't compress as much because when you get a Marshall on 10, it's so compressed already that it doesn't really need it, but cleaner guitars or acoustic guitars, I'll compress. I also got into doing the vocals the same way I do the kick and snare—multing it off and compressing it really hard and sneaking that under the original vocal.

When you say "really hard," how much do you mean?
I would say 10 or 12 dB and at a ratio anywhere from like 4:1 to 8:1. My compression technique is something I actually learned from [engineer] Ed Cherney. He was telling me about compressing the stereo buss when I was assisting him, but I use the same technique on everything. I set the attack as slow as possible and the release as fast as possible so all the transients are getting through and the initial punch is still there, but it releases instantly when the signal drops below threshold. I think that's a lot of the sound of my mixes. It keeps things kinda popping the whole time. Also, you can compress things a little bit more and not have it be as audible.

Do you have an approach to panning?
Yeah, I tend to be a fan of panning things real wide. I'll keep electric guitars, overheads, room mics, and toms hard left and right, and hi-hat all the way to one side. There's not a lot of filling things in between.

The kind of bands I work with want to hit you in the head, so the panning tends to be really extreme. For the most part, they're not really worried about having a Pink Floyd or Steely Dan style mix where everything has its own spot. Also, because radio tends to squash everything back up the middle, I've always found that panning it out like that makes it sound a little bit bigger on radio. If you take the stuff that's panned out wide and make it slightly louder than it should be in stereo, when you listen in mono it really comes together. I find that helps you avoid that all snare and vocal mix thing that you hear a lot of times, and it keeps the guitars up there.

Do you add effects as you go along, or do you get a mix up and then add them?
I'm pretty sparing on effects. Actually, over the last year and a half or two years, I've gradually tried to wean myself off of any digital effects. The last six or so things I mixed, the main vocal effect was a plate reverb and a tape machine or space echo for real tape slap.

Are you delaying the send to the plate?
Depending on the song. Sometimes it works, but with a lot of the music I do the tempos are so fast

that you don't really need to do much delaying because you can't really hear it. It's like the reverb needs to speak right away and then go away. I'm a big fan of the EMT 250 on snare. That's probably been a standard since day one on my mixes. Electric guitars tend to stay dry, and bass is always dry.

How loud do you listen?
Like at conversation volume, probably 85dB or so at the loudest.

Do you usually mix by yourself, or do you have people in the studio with you? Does it matter?
It depends. When we did the *Dookie* record, the whole band was so excited by the whole process (they had never made a real record in a real studio before) that they were there the whole time with their elbows up on the console. On the flip side of that, sometimes the band and/or the producer will come in the first day, and then I won't see them again. I was doing one record where the producer actually left the country and I didn't even know it. About four days into it, I said to the band when they came by to check the mixes, "Should we have the producer come back?" and they're like, "Oh, he's in England." I guess he trusted me.

I like to keep the band involved, and I always put their needs before my ego. I think a problem with a lot of mixers is the ego thing where when the band says, "You know that great sound you have? We want it to sound crappy." You have to take yourself out of it and go, "Well, their name's a lot bigger on the record than mine," so I'll do it and generally the band stays happy.

Chapter 21

Jon Gass

With more than 80 Top 20 hits, 100 Top 40 hits, and more than a hundred gold and platinum albums, Jon Gass has long been the go-to mixer for a credit list that reads like a Who's Who of music superstars, including Madonna, Whitney Houston, Janet Jackson, Celine Dion, Mariah Carey, Mary J. Blige, Usher, Babyface, Earth, Wind & Fire, Lionel Richie, John Mellencamp, and many more. Jon's unsurpassed style and technique have elevated him to a most esteemed position among engineers, working with the best of the best on some of the most creative and demanding music being made today. In this updated interview, Jon describes his method of mixing "in the box."

How much do you mix in the box?
Everything. I haven't touched a real fader in over seven years.
Do you use a controller?
Nope. It doesn't bother me to use a mouse to mix at all.
Can you describe what you do when you first get a track in to mix?
I get so many kinds of projects from all over the world, and some of it is recorded great and some of it is not. Frequently I'll get the stuff that no one else can figure out; the 150 unlabeled tracks kind of thing. Some of it probably couldn't be mixed in the analog world.

First of all I try to get everything organized. A lot of people recording today have never used a track sheet, so they don't understand organization. Their kick might be on track 80 and the snare on 18, so I have to get it to where

I can find everything first. Probably the most frustrating thing about having upwards of 150 tracks to deal with is the different mix pages [in the DAW] that you need to organize things, where on a console you could just reach for a channel and EQ it fast. I do miss that part of working on a desk.

After I know where stuff is, I try to get some basic levels and find out where the problems are. I don't do any fader rides, so if something's not working on a track or a section, I'll split it off onto another channel—that way, I can change the EQ and levels for different parts of the song as I need to. I've had stuff where I'd had to split the lead vocal onto seven or eight tracks so they could all be EQ'd differently because the comp track was from different studios recorded with different mics. Almost every word had a different timbre. It's maddening, and you can't get it to sit in a mix because it's like the frequencies are constantly changing. I still had to automate the EQs just to try to get it to sound similar all the way through. After five or six hours I can finally play through the song through without pulling my hair out. Luckily about half the stuff I do these days I'm producing, and I'm always cutting in mix mode, so it sounds like the record right from the beginning.

Do you still use any outboard gear?

I still do occasionally, mainly something like an Avalon 2055, because I've never found a better EQ. When I first started playing with my rig I couldn't even finish a mix because it just sounded like "the box." I started looking into analog summing and found the Dangerous Music 2-BUS. As soon as I got that, it totally changed everything in that it now sounded like the big analog mix that I was used to. The beauty is that a lot of the hardware effects that I use can now be blended outside of Pro Tools, so it sounds really analog.

Do you have any go-to plugins?
I use a lot of the stock Digi plugs like the basic EQs and compressors because I can use a couple hundred of them if I have to and still have some processing power left over. I like the Oxford EQ too, but for real serious work I'll still throw an Avalon on something. Some of the Waves Guitar 3 stompboxes I love to use as well, even though they're made primarily for guitar.

Are you printing your outboard EQ?
Some stuff I will print if I'm running out of outboard EQs.

Do you have a philosophy about what you're trying to accomplish?
Not really, I just go for it. I'm kind of a musical mixer. I think I try to find the more natural tones of instruments and maybe boost them in that direction as long as it all still fits together. I always think of it as a layer cake, so I just kind of layer the thing.

Can you hear the final product in your head before you start?
Actually, yeah, I can. I know some people push up just the drums and work on them for a while first, but I start with everything in the mix and work on it like that. The reason is that the vocal is going to be in there sooner or later anyway, so you might as well know where it's sitting and what it's doing, and all the instruments are going to be there sooner or later, so you might as well just get used to it. I think that helps me see what I need to do within the first pass, so it doesn't take me long to get a handle on where the mix is heading.

How do you go about building your mix if you have everything in?
I really start searching out the frequencies that are clashing or rubbing against each other; then I work back toward the drums. I try to keep the whole picture in there most of the time as opposed to isolating things too much.

You don't solo things much, then?
Well, I do, but to solo something and EQ it is insane because it's not relative to anything. If there are two or three instruments that are clashing, that's probably where I get more into the solo if I need to hear the whole natural sound of the instrument. I'll try to go more that way with each instrument unless there's a couple that are really clashing; then I'll EQ more aggressively.

I always mix alone now, but when I used to have clients in the room with me I was always afraid to solo stuff in front for them because I think individually the tracks that I mix almost have to

sound bad to work together. It really doesn't matter what it sounds like by itself, because it has to work together with everything else. That's where some of the young producers blow it. They go through and solo tracks and make everything sound fat; then when they put it all together they have a big car wreck.

You're doing mostly EQ cuts? You're not adding anything?
Yeah, I'm definitely into cutting more than adding.

How long do you think it takes you to do a mix?
I don't know that I've done any mix in less than two days. It's usually two to four. I won't take a one-day mix. I can't do it that way.

I usually try to get the rough mix so I know where the client's ears are at, but I don't feel like I'm getting anywhere until my mix begins to smoke it. Usually when I'm finished the rough sounds pretty small by comparison.

How do you use compression?
I try to make it so it doesn't sound like I'm using any. On certain things I'll stomp on it to make it sound compressed, but mostly I'll try to make it so you don't notice it. I don't compress the individual tracks much because if the stuff's EQ'd and layered right, you don't really need to do a ton of compression on the stereo buss. If the thing's laying right, at least with R&B it just kind of sits there.

Are you compressing the master buss inside the box or in analog?
There's some that I'll do in the box and then some outside as well. I have an SSL compressor and a couple of other things that I've used for years. I also compress some of the outputs going to the Dangerous box. It just sounds better that way.

Do you add your effects right from the beginning or do you wait until you have everything balanced and then add them?
I add them as I go. I hardly ever use long halls or long reverbs. I use a lot of effects that are usually set for tight spaces. Sometimes it doesn't sound like I'm using anything, but I might use 20 different reverbs, although not all are actually set for what you think of as reverb. I'm just trying to create more spaces. Though you may not hear it in the mix, you can feel it. If you take them off you'll really miss them, but you don't always notice that they're there.

When you're saying you use tight spaces, are you trying to move stuff back or just put it in its own space?
Yeah, put it in its own space. Sometimes it can be just a chorus, or even a harmonizer with a really short delay time. What it comes down to is I like short, dry sounds.

How short?

Like 25ms or less. I use a lot of 10, 12, 15ms on things. In the R&B stuff, you get a lot of stereo tracks that really aren't stereo. One of the first things I do is to widen the thing out, even if it's only 3, 5, or 10milliseconds, and just get that stuff separated so I can keep my center cleared out. I don't really like that "everything mono" thing.

So what you're trying to do is to make things bigger instead of pushing them back.
Yeah. I think part of that is probably from my early recording days. I didn't really have any reverbs, so I had to use more of the ambiance that was available. That started adding such a new twist as opposed to everything being miked so close and direct all the time. It adds such a great depth to everything.

Do you have an effects template?
If I'm doing an album, I'll try to use more or less the same effects for each song so there's some continuity, but otherwise not really. I do so many different genres and get stuff in from all over the world, so I have to add effects as I go because I'm not always sure where the track is going. I do miss the old days when I had 30 outboard effects plugged up at all times that I either used or didn't depending upon the song.

Are you timing your delays to the tempo of the track?
Depending on the song, yeah. Mainly the eighths, quarters, or sixteenths, but depending on the tune, I'll add in triplets or whatever feels right.

The other thing I like to do with delays is to diffuse them. I'll put a delay through a bunch of stuff just to make it sound worse. Sometimes I'll use Lo-Fi [the Pro Tools native plugin] or something like that just to clip the top and bottom end off and diffuse it off the lead vocal a little bit.

We joke about this guy that mixed a long time ago, and he'd have his delay clearer and brighter and louder than the actual lead vocal. I think that's what kind of got me experimenting with ways to really tone it down.

With all the effects you're using, it sounds like there's a separate one for each instrument.
Absolutely. I very rarely use the same effect on more than one thing.
How often to you replace drums?
I always try to use the drum sounds I'm given and then add what's missing to them. That way it's not completely different from the original drum sound. Drumagog is great for that. I have an 80-gig drive of samples that I've collected over the years.

Do you do a lot of revisions?
Not that many. I don't send a mix for the client to listen to until I'm totally happy with it. Ninety percent of the time there may be no changes at all after I'm finished, and if there are, they're pretty minimal. I just know that when I'm happy with it, they're usually pretty pleased.

What kind of alternative mixes do you normally deliver?
I usually only give them the main mix, the TV mix, and the instrumental.

What speakers are you using?
I mix pretty quietly on the mains, which are Augsbugers with a single 15 and a horn, but you can still feel the power and the punch. I just grew up with them, and I can't use any others. I still have the NS10s and Auratones that I use as well.

Do you have any listening tricks, such as going down the hall or out in the car?
I like to listen outside the room, but one of my favorite tricks is to turn on the vacuum cleaner and lay it up against the wall in the front of the room. Sounds a little strange, but I just kind of want to see if the mix is still cutting through at all. A blender works, too—making margaritas or something. [Laughs]

What percentage of time do you spend working on the mains?
Seventy-five percent or so. I listen pretty quietly, but when starting the mix I'll crank them pretty loud, too. At the end I'll flip around between the different speakers. I'll go really loud on the NS10s and do some adjusting; then
I'll go extremely loud on the big ones and do some more adjusting just to fine-tune things, but I mostly like it quiet on the big ones. I always said I could make a great sounding record on a cassette deck if I just had the right monitors.

You can read Jon's very impressive credits list at jongass.com.

Chapter 22

Don Hahn

Although there are a lot of pretty good engineers around these days, not many have the ability to record a 45- to 100-piece orchestra with the ease of someone who has done it a thousand times. Don Hahn can, and that's because he actually has done it a thousand times. With an unbelievable list of credits that range from television series (such as *Star Trek: The Next Generation*, *Deep Space Nine*, and *Voyager*), to such legends as Count Basie, Barbra Streisand, Chet Atkins, Frank Sinatra, Herb Alpert, Woody Herman, Dionne Warwick, and a host of others (actually 10 pages more), Don has recorded the best of the best. Starting in New York City in 1959 and eventually becoming a VP at the famed A&R Studios there, then later at Hollywood's A&M studios, Don has seen it all and then some. Don's retired now, but his orchestral technique is still the model to emulate.

How is your approach for mixing an orchestra different from when you mix something with a rhythm section?

The approach is totally different because there's no rhythm section, so you shoot for a nice roomy orchestral sound and get it as big as you can get with the number of musicians you have. You start with violins, then violas if you have them, cellos, then basses. You get all that happening and then add woodwinds, French horns, trombones, trumpets, and then percussion and synthesizers.

What happens when you have a rhythm section?

Then the rhythm section starts first. Any time I do a rhythm section, it's like constructing a building. That's your foundation. If you don't build a foundation, the building falls down. I like to shoot for a tight rhythm section that's not too roomy. I think that comes from all the big bands that I did; Woody Herman, Count Basie, Thad and Mel, Maynard Ferguson.

Are you building from the drums or the bass first?

The bass is always first. Everybody relates to the bass. I can remember doing records in New York, and some of the producers would put paper over the meters. I told them, "I don't care, just let me get the bass and I'll balance the whole thing, and it'll come out okay." The only time I can get screwed personally on any date with a rhythm section is if the bass player's late. There's nothing to relate to because everybody relates to the bass player. If he's not there, it doesn't work. Now, orchestrally, the bass players can be late and it doesn't matter because I'm balancing all the other strings and then adding brass and the percussion last. But on a record date with a rhythm section, it's the bass player and the drummer that are the foundation, and the colors come from the keyboards and the guitars.

What's your approach to using effects?

I'll use effects to enhance what I'm doing. A lot of the records that I do are, for lack of a better term, legit records. You can't put a room sound on a drummer on a jazz date. It doesn't work. I've tried it many times, but it ends up like a pop-record rhythm section, and the music doesn't jive

with it. I don't use a lot of effects, especially on the television shows, but I'll use whatever I think is necessary if it's a little dull sounding. If the record doesn't make me bounce up and down, I'm doing something wrong.

Are you panning from the way the conductor's looking at everybody?
No, I do that on movies, but when I'm doing television, I do the high strings in stereo, the low strings in stereo, the synth in stereo, the brass and woodwinds in stereo, the percussion in mono, and anything else in mono. That's a stereo room, and I pan it hard left and hard right.

You don't use much EQ, do you?
I use a little bit. If you use the right microphones, hopefully you don't have to put that much EQ on anything.

How about compression? Aren't you worried about somebody being out of control?
Absolutely not. What are you going to compress in an orchestra?

I assume on a record date it'll be a little different?
Oh, yeah. You might get the French horns jumping right out at you. You might have to put an LA-2A on it and squash them just a little bit, but you shouldn't hear it.

When you were doing the Sinatra dates, I assume it was all live.
Sure, that's the best way to make a record, especially with Sinatra, or Tony Bennett, or Streisand, or any major artist. That's the way they're used to doing it, and it's great. I mean, you really work your butt off, but you feel like you've accomplished something as opposed to sitting there all day and just overdubbing synth pads.

What problems do you have in a situation like that?
Headphones are the biggest problem in the studio. You never have enough separate cue systems to keep everybody happy.

Are you worried about leakage?
No, I try to get the least amount of leakage with as much room as I can. On Streisand, we put the bass player and the drummer in one section of the room with some gobos around, she was in her own booth, three other singers were in another booth, and the whole rest of the studio was filled with great musicians.

How has recording and mixing changed over the years?
Well, just for some perspective, when I started there was no Fender bass and one track only, with no computers or click tracks. Every date used acoustic bass. There was no synthesizer. Bob Moog used to come up to the studio sometimes with his synthesizer that he was working on. It was like 15 feet wide with big old telephone patch cords and tubes, and he'd have us comment on his sounds.

I think some of the problems you have now are that the younger guys don’t go into the studio and listen. You must listen to what’s going on in the studio. Don’t just go into a control room, open faders, and grab EQs. As an engineer you’re supposed to make it sound in the control room like it sounds in the studio, only better. You must listen in the room and hear what it sounds like, especially on acoustic or orchestral dates, and not be afraid to ask composers. Your composers, and especially the musicians, are your best friends because whatever they do reflects on what you’re doing. If they’re not happy, you’re not happy. Remember, the music comes first.

Chapter 23

Andy Johns

Andy Johns needs no introduction because we've been listening to the music that he's been involved in for most of our lives. With credits such as Led Zeppelin, Free, Traffic, Blind Faith, the Rolling Stones, and Van Halen (to name just a few), Andy has set a standard that most mixers are still trying to live up to.

When you're building your mix, where do you start from?

I don't build mixes; I just go, "Here it is." [Laughs heartily] Actually, I start with everything. Most of the people that listen to and tweak one instrument at a time get garbage. You've just got to go through it with the whole thing up, because every sound affects every other sound.

Suppose you're modifying a 12-string acoustic guitar that's in the rhythm section. If you put it up by itself, you might be tempted to put more bottom on it; but the more bottom you put on it, the more bottom it covers up on something else. The same with echo. If you have the drums playing by themselves, you'll hear the echo on them. You put the other instruments in and the echo's gone because the holes are covered up.

Do you have a method for setting levels?

That's all garbage. There was a famous engineer some years ago who said, "I can mix by just looking at the meters." He was obviously an upstart. If you stare at meters long enough, which is what I did for the first 15 years of my career, you find they don't mean anything. It's what's in your soul. You hope that your ears are working with your soul along with your objectivity, but truly you can never be sure.

The only way that you can get a proper mix is if you have a hand in the arrangement, because if you don't, people might play the wrong thing or play in the wrong place. How can you mix that? It's impossible.

The way that I really learned about music is through mixing, because if the bass part is wrong, how can you hold up the bottom end? You learn how to get the bass player to play the right parts so you can actually mix. It's kinda backwards.

I've been into other people's control rooms where you see them working on a horn part on its own. They're playing with delays and echos, and I'm thinking, "What are these people doing?" because when you put the rest of the tracks up, it's totally different and they think that they can fix it by moving some faders up and down. When that happens, they're screwed. About the only thing that should move is the melody and the occasional other part here and there in support of the melody.

Does the fact that you started on four-track affect the way you work now?

Yes, because I learned how to balance things properly to begin with. Nowadays, because you have this luxury of the computer and virtually as many tracks as you want, you don't think that way anymore, but it was a great learning experience having to do it that way.

You know why *Sgt. Pepper* sounds so good? You know why *Are You Experienced* [by the Jimi Hendrix Experience] sounds so good, almost better than what we can do now? Because when you were doing the four to four [bouncing down from one four-track machine to another], you mixed as you went. There was a mix on two tracks of the second four-track machine, and you filled up the open tracks and did the same thing again. Listen to "We Love You" [by the Rolling Stones]. Listen to *Sgt. Pepper*. Listen to "Hole in My Shoe" by Traffic. You mixed as you went along; therefore, after you got the sounds that would fit with each other, all you had to do was adjust the melodies.

What's your approach to using EQ?

You don't get your sound out of a console; you get your sound from the room. You choose the right instruments and the right amplifiers for the track. If you have a guitar sound that's not working with the track properly, you don't use EQ to make it work, you choose another guitar and/or amplifier so it fits better in the track. It might take a day, and it might take four or five different setups, but in the end you don't have to worry about EQ because you made the right acoustic choices while recording.

With drum sounds, even though placing the mics is reasonably important, it's the way you make the drums sound in the room. The sounds come from the instrument and not from the mixer. On rare occasions, if you run into real trouble, maybe you can get away with using a bunch of EQ, but you can fiddle for days and all you'll do is make something that was wrong in the first place just sound different.

How about compression?

I use compression because it's the only way that you can truly modify a sound. Whatever the most predominant frequency is, the more you compress it the more predominant that frequency will be. Suppose the predominant frequencies are 1k to 3kHz. Put a compressor on it, and the bottom end goes away, the top end disappears, and you're left with "Ehhhhh." [Makes a nasal sound] So for me, compressors can modify the sound more than anything else. If it's a bass guitar, you put the compressor before your EQ because if you do it the other way around, you'll lose the top and mids when the compressor emphasizes the spot that you EQ'd. If you compress it first and then add bottom, then you're gonna hear it better.

At what level do you listen?

If I'm listening on small speakers, I've got to turn them up to where they're at the threshold of breaking up but without any distortion, or I listen very quietly. If you turn it way down low, you can hear everything much better. If you turn it as far as it will go before the speakers freak out, then it pumps. In the middle I can't do it. It's just not rock 'n roll to me.

Do you have any listening tricks?
Obviously the idea is to make it work on all systems. You listen on the big speakers, the NS10s, out in the car, plus your own speakers; then you go home and listen again. This is a lot of work, but it's the only way to go.

The thing is that I don't care how close you think you've got it that night, you take it home and play it back in the morning, and every time there are two or three things that you must fix. It's never happened to me where I've come home and said, "That's it." You hear it at home and you jump back down to the studio, and sure enough, you hear what you hadn't noticed before on all the systems there. Every system you listen on, you get more information.

Do you listen in mono much?
No, but I'll tell you this: If you've got a fantastic stereo mix, it will work in mono as well. For example, "Jumpin' Jack Flash" [by The Rolling Stones] is a stereo mix released in mono. People don't listen in mono anymore, but that used to be the big test. It was harder to do, and you had to be a bloody expert to make it work. In the old days we did mono mixes first and then did a quick one for stereo. We'd spend eight hours on the mono mix and half an hour on the stereo.

When do you add effects in the mix?
I have some standard things that I do that more or less always work. I always need a great plate like an EMT 140 and a short 25- to 32-ms delay just in back of the vocal. If it's kind of a mid-tempo tune, then I'll use a longer delay, which you don't hear because it's subliminal. It doesn't always have to be timed to the track; sometimes it can go in the hole so you can hear it. I've been talked out of putting reverb on electric guitars, but Start Me Up has a gorgeous EMT 140 plate on it. Most studios you go into don't even have one anymore.

Do you pre-delay the plate?
Usually but not always. In the old days, like on the Zeppelin stuff, you'll hear very long pre-delays on vocals. You know what that was? That was a 3M tape machine, which was originally designed to do video so it had about a 9-inch gap between the heads as opposed to the 2 1/4-inch gap on a Studer or Ampex. Sometimes I'd even put it at 7 1/2 ips. Another thing we used was the old Binson Echorec (see Figure 23.1). Listen to "When the Levee Breaks." That was me putting two M 160s [Beyer ribbon microphones] on the second floor with no other microphones at all because I wanted to get John Bonham the way he actually sounded. And it worked! Page would say that he made me do it, but he was down at the pub. [Laughs] He did bring me his Binson Echorec for the track, though.

Figure 23.1: Binson Echorec used an electrostatic drum for delay
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Do you prefer analog or digital?

What I like is the sound that's coming into the mixer. I don't want it modified by some tape machine. I've always fought with analog. I've always fought with vinyl. With digital, the sound that's coming in is what you get back. It's much truer than any analog machine ever was. If you have to smooth out your sound with some analog machine, then you're in trouble to start with. With analog, the noise factor is like a security blanket in that the hiss can cover up some weasely things.

But I hate fighting a machine, and I still have to have somebody there with me to help. That's the part of the job that pisses me off. You've now got to be a bloody scientist. Sometimes it makes you too clever for your own good. If you just learn the tune, then you're in tune with the tune. You let it flow through you. Now you might listen to it years later and say, "I think I missed that one." Or, you might go, "I wish I was that guy again. That could not be any better. Who was that man?"

Chapter 24

Bernie Kirsh

Bernie Kirsh has certainly made his mark as one of the top engineers in the world of jazz. From virtually all of Chick Corea's records to winning an engineering Grammy for his work on Quincy Jones's *Back on the Block*, Bernie's recordings have consistently maintained a level of excellence that few can match. Although technical know-how is all important for an engineer these days, Bernie tells us that there are other, more human requirements involved in mixing as well.

Can you hear the final result before you start?

It depends on whether I've tracked it. If it's not something that I've tracked and overdubbed, then I'm discovering it as I'm mixing. Often in the jazz world, it's much more simple because I start out wanting each individual instrument to have a pleasing quality. There's a preconceived notion I have of what that is. If you're talking about straight-ahead jazz, there's a balance that's been accepted as part of the form. In that world, the cymbals are important; the position of the bass, piano, where the horns sit, and all of that kind of stuff has been listened to for decades. It's kind of a traditional form so it's somewhat predefined. If you move away or want to make a variation of that, then you're on your own. If it's something more in the electric vein and something that I've worked on, then I'll come up with a notion of where I want it to go.

Where do you build your mix from?

The first thing that I actually look for is the melody. After that, I'll go for the bottom of the mix, usually the bass. I don't necessarily go for the drums first. Before I hit the rhythm I usually try to get the melody and some sort of harmonic setup first, because I want that to be clear, and I'll often shape the rhythm to accommodate that. That's the simplicity of it. If it's something that's more hard-hitting, I'll spend more time with the rhythm to get those guys pumping together.

Do you have certain frequencies that you seem to come back to that need attention on certain instruments?

Let's say for piano (which I've dealt with a lot), typically what happens is that in the analog domain it loses definition and openness if it's mixed some time after it's been recorded, so I'll usually boost in a couple of areas. First, up around 15k (sometimes that gets lowered down to 10 or 12, depending on the instrument), and maybe a little midrange at 3k or 5k. It depends on the instrument and setting, but that's pretty typical. I'll do the same thing usually with cymbals. I'll add between 12k and 15k on cymbals pretty typically. Those are the normal areas of EQ that I find that I'm constantly using.

The frequencies that you adjust seem to be a little different from other genres.

With this kind of music it's all about trying to go for more of a natural sound, for lack of a better phrase, so if there's going to be any hype at all, it's going to be with the loudness button (on a stereo receiver) where you get the larger bottom and accentuate the top.

Normally, if you're going to add anything else to a piano, for instance, you're in the 500Hz range adding some warmth. I sometimes find that when I finally get to mastering, the mastering engineer wants to take some of the warmth out for clarity purposes with just a little notch around 200 or 300. My tendency is to go for the warmth and then sometimes take some of that back out to achieve a little more definition or clarity later, if needed.

Do you have an approach to panning?
No, I normally keep things wide, drums in stereo and piano open. I personally like a wide piano. I like it so that it feels like you're sitting at the instrument.

You do it wide, left to right?
Yeah, wide left to right. I position everything as the player is seeing it rather than the audience, so the drums are from the drummer's perspective, piano is from the pianist's perspective, etc., unless there's a leakage situation where I have to worry about the phase. If, for instance, the piano and the drums are in the same room, I have to make sure that the cymbal is appearing in the right place and isn't smearing because of the leakage into the piano.

Is there a certain psychology that you use when recording?
I wouldn't call it psychology, but it's in the realm of human interaction. I think there are certain basic things that occur in that little microcosm called a studio, which a lot of guys don't recognize. You're getting into some basic human sensibilities that may not be apparent as you look at it. For instance, you have artistic creation going on. You have a guy who has come into the room who has done something that's very, very close to who he is. It's not PR. It's not show. It's something that he holds very, very dear to himself. Now he's, for lack of a better word, open and vulnerable, and he's not being social.

So now you've got an engineer in the room whose attention isn't on that. Often you get engineers who, through various different bits of behavior, will invalidate the artist, evaluate the artist, and not respect the frame of mind that the artist is in when wanting to make his musical statements. In other words, not looking at what the artist is doing at the moment. I think you'll find that the best engineers, the ones that the artists want to work with, have a notion that what the artist is doing is important and is something that needs to be treated with attention and respect. When I say that, I mean not to hold it up on a pedestal, but to understand that the action is something that's very close to the artist and not just a commodity.

For some reason, the creative process is different in the jazz world. Guys are coming in, not necessarily to just lay down just a rhythm track, but with the idea of making music. Because of that I put a lot of attention on making the players happy with what they're hearing and making it comfortable for them. I don't work with a lot of engineers so I don't see it, but from the feedback I get, a lot of the younger guys don't recognize that element is really important. It seems like the job is really 10 percent technical. The rest of it is how you work with people and help them get what

they want.

Chapter 25

Nathaniel Kunkel

It can be said that Nathaniel Kunkel was always meant to be an engineer. The son of session drummer extraordinaire Russ Kunkel, he grew up literally at the feet of the best that the LA music scene had to offer.

But nepotism doesn't mean much in this business, and Nathanial had to work hard and pay his dues just like thousands of others, working first as an assistant at Jackson Browne's studio and then with George Massenburg. All that has paid off handsomely, as Nate is now one of the most in-demand mixers in the business, with credits that range from James Taylor, Lionel Ritchie, and Sting to Good Charlotte, Fuel, and Insane Clown Posse. Find out more about Nathaniel at studiowithoutwalls.com.

Do you start a mix with a particular approach?

I think I learned the best approach on how to begin a mix from Ed Cherney. Ed just sort of pushes up the faders and listens to the song. Maybe he'll just pull the guitar down, and maybe he'll just push up the bass and drums and vocal and listen to it. He really just spends a lot of time listening, and he really gets a feeling for where the gems of the track lie. In listening to all the stripped-down versions, he finds the little moments that can be brought to fruition in every one of the tracks. Then he does it like everyone else, where he pushes up the kick drum and checks the phase with the overheads, then puts the bass in, and so forth. Everyone kind of does it the same way, but it's really what you are looking for out of the individual instruments.

Can you hear the finished product in your head before you start?

Absolutely. I could hear the final mix in my head for a lot longer than I had the skill to get it there. What happens is that when your skill lets you down you run out of time, which doesn't really mean that you have to leave the studio, but that you no longer have the same perspective as when you had it fresh in your head. To me, the secret combo is for you to come in, have a really great idea about what you want the song to be, then have the skill set to get the tracks to that point before you lose perspective because you're fatigued.

So how long does it usually take you?

I usually lose perspective in about an hour and a half or two hours. I can mix solid for about eight, but I find that I can't do really vigorous knob-twisting after about an hour and a half into it; then I just start to chase my tail a little bit.

What do you do after an hour and a half? Do you switch to another song?

Yeah, unless I can do something else to break the concentration for a half hour or so.

Where do you start your mix from?

If the song is a consistent dynamic all the way through, like a rock song, I really don't have a

method. I just sort of push it up. If I've gotten a great balance on the guitars and vocals before the drums, I'll just mute the guitars and get my drum balance a bit better, then just group it and balance it against the guitars and vocals. If I've gotten something really great going on, like a background blend, I'm certainly not going to whack it in order to get a better kick drum sound. Because if it's something consistent, like a rock song, you certainly don't get into much trouble, because you can push stuff up or down as you go along and sort of end up in the right place.

In situations where there's an intro where things are quieter, like an acoustic and electric guitar and a vocal before the drums kick in maybe in the chorus, you don't want the vocal level in the chorus to change very much, but you want what's happening between the vocal and the drums to be right. That balance is very difficult to build after you've built an introduction, so often I'll go and get a drum sound and push up a vocal and get a rough blend with the band, and then I'll go back to the intro and I'll push up the other instruments around that admittedly loud vocal. I'll build it so my automation brings it back to the balance that I had before, so if I have a song that will dip down dynamically once or many times, I'll go and get the core instruments to sit exactly where I want them in the loudest place. I'd rather have the automation return it to the up part rather than the down part.

What's the difference for you between mixing on a console and mixing in the box?
There's nothing different when mixing in the box that I didn't do when I was mixing on consoles and tape machines. It just takes me less time. I have to mix in the box. I couldn't afford the infrastructure necessary for the sheer quantity of tracks I have to deal with in the projects that I do.

Did you have to change your approach when you decided to mix in the box?
I find that I'm doing it exactly the way that I used to on a console. I find that I use my console [an Avid Icon controller] a lot more like an analog console. I put the same EQ and the same compressor on every channel so I never have to go to the plugin menu. I still use all outboard vocal compressors, my Distressors, GMLs, and Alan Smarts.

I also use tons of headroom. When was the last time you took a console that clipped at +25 and ran it at +24? You just don't do that in analog, so why would you do that in digital? I use at least 10 or 15dB of headroom on my buss. If I'm going to print a loud version, I'll take it out to an [TC Electronic] M6000 or something that does a really outstanding job of handling over-levels and then bring it back into Pro Tools and not change it.

When you used to sit down at a console with a tape machine that was overbiased, you would play it and say, "Something doesn't sound right," and then you'd turn around and address it. How many times have we sat down behind a Pro Tools rig and said, "Ug, something's wrong"? That's the same thing as with a tape machine; it's just a different toolbox. It may be the headroom or clock distribution or a variety of things, but you have to pay attention and then fix the things that are wrong. If it sounds digital, then try something different. It's a new toolbox, but it's really the same

auditory skill set.

Do you have any tricks that you use when mixing in a DAW?
No. That's frightening to say, I know. Maybe it's because I was taught by fantastic analog engineers. So what can you really do with audio? You can store it, you can change its level, you can delay it, or you can EQ it. That's really it. Reverbs are combinations of two. So mixing inside the box, what in the world is different about that? It's still the same problem that we've been having forever—making good artistic decisions and then following through with the right toolset. So for me there's nothing that I do within a digital workstation that I didn't do in the analog domain before. I can do it better in some cases, but it's not a different task.

Do you have an effect that you keep coming back to?
My M6000 is my effects processor of choice. It's the single best effects processor that I have ever seen or used, but never underestimate the power of a good delay. Delays are really wonderful ways to open up mixes. Maybe it's just a simple ping-pong or a guitar on one side with a delay on the other; they don't have to be loud and obvious. They can be subliminal and add some groove.

Effects in general don't have to be obvious. The best effect to me is when you're not really aware of it. I often have rock songs that have all kinds of things going on, but when everything is raging you don't even hear it. All you'll know is that it has a little more swing than it did before. That to me is the win of using effects. When you can just enhance the emotional response that people have to the music without drawing their attention to some kind of trickery.

How much compression do you use?
I don't want to say. [Sheepishly] Sometimes on a vocal I may use more than 10dB. There are times when there's singing when it's not in compression at all, but if my limiter hits 15 or 20dB of compression and I don't hear it, I don't think about it for an instant more.

Do you put a compressor across the stereo buss?
I use very little. I find I'm using more multi-band compression on my buss as nothing more than a way to elevate my level so I have something that's competitive for approvals, but I print almost all of my mixes without limiting.

The truth is that I compress things enough instrument-wise, so I don't really need to compress more on the buss. When I do compress the buss it's maybe 2 or 3 dB. The multi-band allows me to get a more competitive level with less compression.

Do you have any special listening tricks?
I listen quietly as much as I can. It's hard to check the kick drum level when it's quiet, so certainly you have to push it up every once in a while, but I fatigue pretty quickly when listening at loud levels. I can make better emotional and timbre decisions before I fatigue.

What's your approach to mixing in surround?
I guess if I were to encapsulate the rule, the things that I used to put in the middle I put everywhere now. Bass, kick drum, snare drum, lead vocal—all the stuff that has a lot of mono correlated information goes a bit to every speaker, except maybe the center. If I put something in the front, I will very rarely put it in the center and the left and the right. I will put it in the center and the surrounds if I want to pull it more into the middle of the room. If I want something off to the side of the room I'll go Left, Right, and Right Surround so it leans to that side.

What do you usually put in the center?
That changes. I always put something in the center. Mostly vocal.

How about the LFE?
Jeff Levison from DTS told me early on, "Dude, here's what you have to understand. The LFE is a low-frequency effects track. It's used when you run out of low-frequency headroom in your other channels." That was in the very beginning when I was using tons of LFE, and we'd go into these rooms with an improperly aligned sub, and the low end for the entire track would be wrong. Then I started mixing with bass management, so I only go to the 0.1 when I cannot put any more level on the main channels and I want more bass.

One last thing. What was the most important thing that George [Massenburg] taught you?
After about five months of working for him, he said, "You know, Nathaniel, the hard part about this gig is not getting good sounds or getting along with the artist, it's about paying attention to every little thing that's done every moment of the day and knowing exactly what it means." The really great engineers know what will happen with every button in the room, and that's why we all go so bananas when people change things.

Chapter 26

George Massenburg

From designing the industry's most heralded audio tools to engineering classics by Little Feat, Earth, Wind & Fire, and Linda Ronstadt (to name only a few), George Massenburg needs no introduction to anyone even remotely connected to the music or audio business. You can find out more about George and his GML audio gear at massenburg.com.

Can you hear the final mix in your head before you start?

No, I generally look for a trace of feeling and I diddle things until I get a response. Whether it's EQing or changing arrangements, it's got to work as a feeling. And as such, I feel that what I do is significantly different from anybody else. I don't go into a studio to make money. [Laughs] I go in to experiment.

Is that a collective feeling or is it singular?

Just about any successful piece of music is not something that can be performed by one person. It's almost always a collaboration. I can't think of anything that only one person has done in pop music.

When you begin to build your mix, where do you build it from?

I always start rock and roll with drums, but very quickly I'll get a voice in there so that the instruments are crafted to work to the texture and the dynamics of the voice. I don't have any real rule. I actually can start just about anywhere.

When you start with your drums, are you starting with the overheads first and building around that?

Yeah, I generally will start with overheads.

Room mics or overheads?

Well, first and foremost I'm listening to the music, so I'll start with whatever gives me the best picture of what's going on in the room. I'll get a fast, overall mix and while I'm figuring out the tune, I'll start listening for problems or things to improve. The problems might range from a less-than-effective instrument amp or a mic placement to some big, funny boink somewhere that's sticking out. I like to tune things and line up overtones. I feel that equalizers are best used when used the least. I use them mostly to get rid of tones that are somehow not flattering. I'll most often use parametrics, sharp and subtractive, to look for the two or three biggest out-of-sorts characteristics. A snare drum, for instance, has any number of boinks that I'll locate, and I may take them out or bring them up as I'm listening to the whole presentation, but I'll already know what and where they are.

Do you have an approach to using effects?

I don't have an approach. This is probably my biggest strength and my biggest weakness at the

same time. I really try to invent everything from scratch every time I walk in. But yeah, I have basic things that I keep going to.

When you're beginning to set up for a mix, are there certain effects that you automatically put up?
I'll have about eight delays set up. If I can send something into a delay, I'll do that because it takes up a lot less room. If I can make it sound like a reverb, I'll use it. I'll always go with the delay instead of a reverb if I can hide it.

Hide it meaning time it to the track?
Yeah, timing it so you don't really hear it as blatantly. You hear richness and warmth.
And the timing is what? Eighth notes or dotted or triplets?
It's musical, and the timing will change. Often it's just by feel. I just put it up and try to get something that rocks.

What's your approach to panning?
I've got two or three different approaches, and I'm always changing it. I used to be impressed by a drummer liking what I did, so I pretty much only got a drum perspective, but I've gone wide and I've gone narrow.

Do you have a general mixing approach?
I want to hear something authentic. I want to hear an authentic room or an authentic performance. I want to hear authentic instruments. It's not necessarily a sophisticated or elegant thing. It's just authentic. In stereo I try to paint a picture that makes sense—that your brain doesn't say, "Hey, what are you trying to put across on me?"

How are you applying compression during the mix?
The big difference between engineers today is the use of the compressor. At one time or another, I tried to compress everything because I was building a compressor and I wanted to see how it did on every instrument. I'm a little off compression now because there are so many people who overuse it. Everything is squeezed to death. As a result I'm backing off. When anybody goes that far out, I'll go the opposite way as hard as I can. Generally I will pretty much always have an option to compress the mix. I'll use my EQ, my compressor, then my converter and an M5000 to do three-band; then I can dial it up from extremely subtle to pressed ham under glass.

I'll always compress vocals. I may recompress vocals again during the mix. I'll almost always have a bunch of compressors if I have to bring an element or a group of elements together like a background vocal, level them, then drop them into a pocket.

Then I'll do some extreme stuff like compressing a room and then gating it. Maybe I'll compress a drum room and then gate it with the snare drum to get a real rectangular reverb. I do that a lot. Maybe I'll add reverb to a guitar and then gate the result of that. I do that some. Boy, I wish I

could give you a rule.

What are you trying to accomplish?
Trying to get a thrill. [Laughs] I'm almost always trying to get, as Lowell George used to call it, "decibel excursion," which is a BS term but I love it. I try to make an instrument denser or give it some weight. Half of it's from reverb or ambiance, and the other half is bringing that ambiance right up in your face, which is compression.

How about monitoring? What's your typical monitoring setup?
I monitor on a lot of different things. I might go up to the wall monitors to try to hear subsonics. I'll go to Yamahas to hear what the idiots at the record companies are listening to. Tannoys for fun. KRKs for fun. Headphones.

You listen on headphones?
I listen on headphones because you can hear if you're making a mistake. [Bassist] Jimmy Johnson taught me that. He would always find that snap in bar 30 of the sax solo, and you'd listen to it and sure enough, a tiny little snap to get rid of. For the kind of music he was doing, that was appropriate.

What levels do you usually monitor at?
Everything. I'll monitor way loud to see what rocks. I'll monitor at a nominal level to get sounds together; then I'll monitor about 5dB over background noise to hear all the elements in focus. If a mix works at 30dB SPL, 2 dB SPL, it'll almost always work a lot louder.

What are you listening for down that low?
How the instruments work together and to make sure that you don't lose anything. If you can hear everything at that low a level, then when you turn it up you'll have a very even balance. That's the way to get everything in the same plane, by listening extremely low.

Do you have any playback tricks? Do you go outside in the lounge and listen through the walls sometimes?
All the time. I'm a big one for hallway. I hate cars. Through the control room doors is always an important thing for me, because I almost never do loud playbacks. I like listening around the corner and on a blaster.

How many versions of a mix do you normally do?
I believe in one mix, and I believe that either it's right or it's not right. I will walk out of the control room with only one mix because at that point, it's important for you to let go of it. It doesn't belong to you anymore.

Do you go back often and do any touchups or remixes?
Yeah, all the time, but I think usually we go from scratch again. Some of the best mixes I've done

have been the fourth or fifth or sixth passes. I remember on “Shining Star” [Earth, Wind & Fire’s hit], we kept going back in the studio and tracking, and I think the more we went back in, the more we found what didn’t work.

It seems like it’s so much easier to refine things as you go back like that.
Oh yeah, because you know what your priorities are. You know what doesn’t work, because the first couple times you go in, you’re trying exotic EQing and delays. You go back in, and it doesn’t make any difference except for the stuff that’s right. The thing that makes a difference is vocals. I’ve spent more time than anything else trying to find how to do vocals and how they tell the story.

What you bring to the table in the control room seems to come through. I’ve been gifted to work with great musicians, and any of the sounds that we get—any of the sounds that any of the really good cats get—it’s because of great musicians.

Chapter 27

Robert Orton

If you've enjoyed the big hits from Lady Gaga's *The Fame* and *The Fame Monster* albums, such as "Poker Face," "Paparazzi," and "Just Dance," then Robert Orton is your man. After spending eight years at the side of producer extraordinaire Trevor Horn, Robert has gone on to craft hits for Robbie Williams, Robin Thicke, Carly Rae Jepsen, Flo Rida, Kelis, Usher, Mary J. Blige, and Marilyn Manson, among many others, while winning a few Grammy awards for his work along the way. Robert is also one of the first hitmakers influenced by earlier versions of this book. Says Robert, "The Mixing Engineer's Handbook is one that I remember studying very closely when I was an aspiring mixer hoping to catch a break." Find out more about Robert at hitmixer.com.

Can you hear the finished mix in your head before you begin?

I think that having a pretty clear vision of where you want to go from the beginning is one of the most important things in achieving a good mix. Having said that, I don't always work that way. Sometimes I just see where the mix takes me, but a lot of the time I'll immediately think, "I know what this needs," and have a fairly clear idea of how I want it to sound. It's all about being bold and making big decisions.

Are you mixing mostly in the box or on a console?

I'm mostly in the box in Pro Tools with a load of plugins. I've got a few bits of outboard gear as well, mainly a bit of compression, like some [Empirical Labs EL8] Distressors, because there's not much that sounds like them. Sometimes you just need a bit of analog to find the sound you're looking for, but for the most part I'm in the box. I guess I've always really mixed that way. I like the way it sounds, and it's so much more convenient when it comes to recalls.

Are you using an analog mix buss?

No, I've tried those kinds of things, and I haven't felt that I've gained a lot from using them. Basically I mix in the box, and then I've got a couple of things that might go over the mix, but I don't buss it out or stem it out or anything like that.

Do you use a controller?

No, I just use a trackball. The few times when I mixed on an [Avid] Icon or that sort of thing in the past, I've found that I ignored it most of the time. I'm so used to doing rides on the trackball that I forget that the controller's there. I have a really simple little setup. I just sit in front of a pair of monitors with a computer in a well-treated room.

Where do you start your mix from?

I start with all the faders up. The first thing I do is listen to the rough mix of the song I've been sent a few times to get the idea of what the producer and artist intended. Then I just push all the faders up. I like to mix while hearing everything. Once I've got a balance that feels about right, then I might listen to the drums and see how they're fitting with the bass and see how the vocals

are sitting on top of that, but I wouldn't say that I build a mix up by adding one instrument at a time.

Do you augment the drums, or is what you're given what you work with?
I try to work with what I'm given just because augmenting can change the character unintentionally if you're not careful. Having said that, sometimes you get a kick and think, "This is never going to sound how I want it," so you have to add something in. Of course, that happens more with something like a rock band.

There always seems to be a lot of layers in your mixes. I know you started with Trevor Horn, and his stuff is like that as well.

Is that where you got that technique from?
I worked with Trevor for eight years, and I learned so much. He's so talented and just an unbelievable guy. I learned something from him every day I worked for him, and I'm sure that if I worked with him again, I'd still learn something new from him. It's true that he layers a lot of things. Most of his multi-tracks tended to have a lot of parts, and mixing that stuff was difficult in terms of the track density, so I got used to layering things and figuring out how they fit together. That rubbed off on me, and I certainly do approach mixes like that now.

You're not afraid to go wet on a mix even though the trend is to stay dry.
I just do what feels right, really. It's about creating shifts in perspective and excitement between different sections of the song. There's definitely a lot of that going on in the Gaga stuff, with extra delays that come in on the chorus and that kind of thing. Dry is good because it can be in your face, but a bit of wetness can position things and stop them from bumping into one another.

You seem to use a lot more delay than reverb.
Yeah, I don't use much reverb at all, actually. Sometimes I'll use it more as an effect if there's a big moment that needs some sort of emphasis, but generally I try to stick with delays because you're left with more space in the track.

Do you use a set number of delays right from the beginning of the mix?
I do have a template that I start from, but to be honest, everything's individually tailored for each mix. I don't really know what I'm going to need until I start diving in. I normally just listen to it and tweak as it's needed. I spend a lot of time on the vocal because that's often the most important element of a mix. I add effects while listening to the vocal in the track, because what's on a vocal can really influence the way the backing tracks feel against it.

I notice that you use a lot of stereo delays as well.
Yeah, I do that to create excitement in the mix. You've got frequency to use and you've got depth to use, but certainly there's width that you can create excitement with, too. Things that sit far left and right in the mix can make it sound more exciting. Having different delays on either side can

often make a vocal sound bigger, but at the same time they don't get in the way of the direct sound.

Do you have any plugins that you almost always use?
Yeah, I do. It's funny; you do tend to go toward the same things that you like the sound of. I love all of the Waves plugins, especially the modeling ones that they've been doing lately. The Chris Lord Alge compressors sound great, and I use the Neve and the API EQs a lot. I use [SoundToys] EchoBoy a lot for delays, and the Bomb Factory Moogerfooger is really useful.

I try to push myself to use the plugins in creative ways instead of the obvious ways that they're always used. For problem-solving I tend to go toward plugins that I like the sound of and I know what they're going to do, but sometimes you're looking for a vibe when something's not happening in the parts that you're given. That's when I'll look for something more creative with a plugin, like an amp model on a vocal or an automated sweeping filter over the backing track. The sort of thing that completely changes the sound and creates something new for you to feature or lock onto in the mix.

Do you have a particular approach to using EQ?
I use a number of approaches, actually. Sometimes I'll hear a frequency that I don't like; then I'll use some subtraction. Other times I might think, "This just needs to be brighter," and I'll just turn the knobs until it sounds right. I don't think too much about the frequencies, and I'm definitely not afraid to boost a lot.

Having said that, I don't just go in and EQ everything. I think the best advice I could give anyone who wants to mix is to learn when not to process something. The more you process things, the worse it sounds, I generally find. To me, mixing is more about balance and groove and getting that to happen. I always try to get the mix to sound as good as I can just with the balance before I go in and start EQing. I might fix something at the beginning of a mix if it feels wrong to me, but I don't just jump right in and EQ things. I think that leads to more of a mess than anything.

How much compression do you use?
I'm quite light with it. I do often use some compression over the mix buss, but I don't automatically put it on. I find that sometimes compression helps the groove of the song. I don't really use compression to hold things down in their place unless there's an obvious problem. I might compress a lead vocal quite heavily to maybe bring out a certain energy in it, but generally I don't compress a lot. A lot of mixing for me is about feeling the movement and the groove, and a little bit of compression in the right area can help certain elements fit in the pocket and make that happen, so I might use a little for that.

The Gaga stuff is a really good example. "Just Dance" has a slightly odd drum pattern that sounds a bit like four on the floor, but the four was actually just the snare. I used an SSL buss compressor across the mix, and it really helped that feel because of the way the kick and the snare reacted to it. That's a good example of how buss compression can influence the groove of a song. I use

compression more like that, really. I don't automatically reach for it, but sometimes it can really help accentuate the groove.

What's your approach when you have a song that uses a synth instead of a bass guitar for the bottom end?
I think you have to approach it slightly differently, because quite often those types of synth sounds interact with other parts of the song. Maybe the top end of it is buzzy or something like that, and it's not always just the traditional low end that you're looking for from some of those parts. Sometimes I approach that by maybe multing it out so I've got the bottom end from it [on one channel], but treat it in parallel and then blend the sounds together.

I've noticed that in those kinds of songs you've made the kick a little bigger than it might normally be, so it takes up a lot of those frequencies as well.
I tend to mix the kick drum quite loud in that style of music. I know that's a bit of a trend at the moment as well, but I like to hear it thump you in the chest. Sometimes I'll use a trick like compressing the bass every time the kick hits so you don't need to balance the kick as loud for it to have impact.

So you're keying it, then?
Sometimes. That's certainly a good trick when you have a dense bass synth that's taking up a lot of space. Sometimes that kind of trick can open up space in the mix if done subtly enough.

How long does it take you to do a mix?
I do a mix in a day. I normally start at noon and keep working until it's done. Sometimes I go home at 10 in the evening, and sometimes I go home at four in the morning; it depends. When I'm not under a really strict deadline, I like to come in and listen the next day as well. Going away and getting a fresh perspective with a new pair of ears can be very valuable.

What monitors are you using?
I'm using some ProAc Studio 100s, which is what I listen to most of the time. I also have some Quested VS2108 midfield speakers. I also love listening on a laptop just to hear how it sounds on something really small, which is the kind of thing that people might listen to music on a lot these days.

Do you listen on earbuds?
No, I don't. I do have a pair of Sennheiser headphones that I really like. I use them right at the end of a mix just to be sure that I haven't missed any clicks or anything like that. Sometimes things jump out at you a little more obviously on headphones.

How loud do you mix?
I monitor really quietly most of the time. When I'm getting the drum sound and again toward the end of the mix,

I might give it a little blast on the bigs to just kind of check the bottom end. I might turn it up when I'm doing the vocal to see if there are any frequencies that catch my ear, because sometimes when you turn it up the sibilance will irritate your ear a bit more.

There's only a few times during the mix when I turn it up, though. The rest of the time I monitor really quietly. If there's someone else in the room tapping on their computer, it bugs my ears. [Laughs]

How many versions of a mix do you do?
Unless someone specifically asks for a vocal up or vocal down, I normally just print the mix, an instrumental, a TV mix, an a cappella, and some stems. I kind of prefer it when the mix I give them is the definitive version and there's no question about it.

How much do you automate things?
I do quite a lot of automation. I spend the first part of the mix getting a rough balance, but I spend the second half of the mix just riding everything. Balance is certainly a dynamic thing. I might change the feedback on a delay or alter the filters on a section or ride the effects sends, and I certainly do a lot of volume rides.

Are you drawing it in or using the trackball?
Mostly with a trackball. I do draw in some things early in the mix, because maybe one sound might be too loud or too quiet in a certain section, so I do broad brushstrokes at that point. As the mix is coming together, I'm doing a load of rides with the trackball just by feel.

I mix in quite an odd way, I think. I sort of listen to each section of a song repeatedly. I might listen to how the pre- chorus goes into the chorus, and I'll go over that one section quite a lot of times and push different things up, riding them with the trackball until it feels musically right; then I'll go to the next section. I'll sort of work my way through the song like that rather than doing full passes from beginning to end. One of the reasons is I'll think, "I've got to go back and fix that thing I just heard." If I let it run to the end of the song, I'll have forgotten about it. It's a good way to focus myself.

What have you learned between the time you just started to mix full time and now?
I don't think I can pick one thing. I think the amazing thing about mixing and the thing I love the most about music in general is that every day I learn something. I'm always surprised by how sounds interact with one another, and if you change one thing it might change something that you didn't expect. The way that music works and the way we perceive it is a constant amazement to me. I feel like I learn something new every single day.

Chapter 28

Greg Penny

Born into a music business family to bandleader/producer Hank Penny and hit recording artist Sue Thompson, Surround Music Award winner Greg Penny seemed destined for a life in the studio. Indeed Greg's production aspirations resulted in hits with k.d. lang, Cher, and Paul Young, among others, but a meeting with Elton John while Greg was in his teens turned into an award-winning mixing journey with the legend many years down the road. Greg gives us an inside look at his long relationship with Sir Elton as well as some insight into his sensational surround remixes.

How did you get connected with Elton?
I met Elton backstage at his first Las Vegas gig, and then the second time he came through I brought my Mom down to see him. He was a fan of hers, so he was very happy to see us. After that I stayed in pretty close contact with him, and I'd go to his gigs when he'd come through town, so he knew that I was a real sincere fan and that I wanted to be in the record-making business.

When I was 17, I decided to go to Europe, and Elton invited me to stop by the Chateau in France to check out what they were recording, which was Goodbye Yellow Brick Road. The day I got there, Dee [Murray—Elton's longtime bass player] was fixing the bass track to "Saturday Night's Alright for Fighting." They cut the track the night before, and he was just patching some things up.

That's an awesome education right there.
It also had a lot to do with me inheriting Gus Dudgeon's [Elton's producer] projects because I had so much respect for him and he was one of my heroes. Gus let me sit in on stuff and never shooed me away. He never had an attitude with me and was always encouraging because he knew that this was what I wanted to do, so he gave me the space to hang out and be a fly on the wall. I didn't go to London to the mix sessions for the record, but I stayed pretty up to the minute on everything they were doing.

Over time we've remained friends; then in the early '90s I produced these records with k.d. lang that were real successful, and it turned out that Elton was a fan of them. He rang me one day and said, "I want to do this album of duets. Would you do a track with me and k.d.?" At the tracking session he asked me if I'd do more songs on the duets album, which I did, and then he asked me to do his next studio album, which was the Made in England album that became really successful.

That kind of got me to where all I did was Elton stuff for about three years, from studio to live stuff to remixing other things to following up on B sides and things for charity albums. There was just a lot of work, and I gladly fell into it and kept on going. A couple of years ago I went back to him with the idea of mixing his catalog in surround, and he was really open to the idea.

Do you incorporate the mixing tricks you learned from Gus now on your more recent stuff?
Yeah, more often than not I do because I find that when I listen to mixes where I've been subtle or

timid about the level of a part, I tend to later feel that more dramatic colors or more dramatic mix rides had to be done. I've gotten more radical with things, and when it's time for them to be featured, I'll crank 'em up.

Let's talk a little about your general mixing philosophy.
I usually start a mix from the vocal. Obviously there are instrumental things that you have to be concerned with, but my thought is how the vocal is going to work with the elements that are supporting them, so I guess I mix from the vocal out.

Sometimes I'll poke around on drums and bass to get a good floor and then I'll take them out and put the vocal in to try to get a really good sound on it; then put all that together with all the other elements around it, always bearing in mind that the vocal has to be the thing that pulls people in. For me, that's really the most important thing, although that's not necessarily true for everyone who mixes or all artists. There are grooves from each artist that are special to them, but they're nothing without their vocals.

It's been fun to do the surround thing because I've been able to highlight things just by virtue of the fact that in stereo things get crowded. In surround I've been able to pull out instrument passages around the vocals.

What's your approach to mixing surround?
It really started with *Goodbye Yellow Brick Road* because I had a few weeks to sit with it and come up with a formula. I thought, "Here's this incredible record that's got tons of guitar and keyboards on it, so why don't I put the vocal in the center monitor most of the time, and the only other things that enter into that monitor are double vocals or harmonies or maybe even a solo instrument? Then I'll bleed out a little bit of the center vocal into the left and right fronts, so if Uncle Bob comes over to the house and sits at that end of the couch he's not missing the lead vocal. Then I'll use divergence and spill a little of that lead vocal into the rear monitors also for that purpose. Or we could give Elton the front of the room for his piano and Davey the rear of the room for his guitars, or even put them in quad like in 'Saturday Night's Alright for Fighting'—four guitars, four speakers, with each guitar getting its own speaker." That started me moving in that direction, and then it got more radical as I went through the album. By the time I got to *Madman Across the Water* with those incredible string arrangements by Paul Buckmaster, it was amazing to put the whole string section in the back and the band in the front. They recorded everything at the same time, so when you listen to it it's like you're sitting midway between the band and the string players, just like on the original date.

It sounds like the fact that everything was cut live greatly influenced your surround mixing philosophy.
As Nate Kunkel says, you've got to make records for the guys who have the right system, because if you back off from that and try to dilute the vision down to the most common denominator, we'll never get anywhere. You have to somehow set the benchmark. More often than not on these blogs

on the Internet about Elton's mixes, I either get slammed for being sacrilegious or lauded as a genius. They always talk about how the use of the surrounds is extreme on Elton's records, but then a lot of guys say, "Finally, I'm able to hear my system because I've got some stuff that puts something through all of my speakers." My objective is to use the system entirely and not to be too timid, but there are some songs where you just don't have enough data to put in all the speakers and have it make sense without it seeming lopsided.

What kind of gear are you using?
I use a Pro Tools HD system. All of Elton's stuff is at 24/96. It seemed that the jump up to 96k was much more noticeable than up to 192k. I have a very simple single interface with a [TC Electronic] System 6000. I mix in the box. On occasion I'll go out to a little TL Audio tube desk to try to get analog warmth out of some things if I can't get it within the box.

For speakers I use Dynaudio AIR6's. I come right out of the back of the HD interface right into the back of the speakers with nothing in the way because the controller that comes with it is sufficient for me. I'm not doing any downmixes with the mixes that I'm doing, and if I need that I can test it with a Waves plugin across the buss rather than having something like a StudioComm box [surround monitor controller]. The Dynaudios are incredibly accurate right down to a whisper level and then can also get incredibly loud. The system also translates very well when my mixes are played back in other rooms.

Are there any tricks you've found to make mixing in the box sound better for you?
I use a pretty simple set of tools inside. I don't use a lot of on-board plugin reverbs. I have a couple of short things that I use, but predominantly I use the System 6000 surround reverbs. If I don't have enough firepower with the one unit, then I'll print the reverbs and go back and do it again.

I use the Waves 360 suite a lot. I use their panner because I find it's easier to automate when I can just grab the toggle thing and fly it around. I also like the L360 limiter.
It's actually a very simple set of tools that I use. I don't believe for the albums I'm doing that I need to get complicated, because when these albums were originally mixed they only had a couple of plates available, maybe one that was a second or so long and then another one maybe about two and a half seconds. Over time they got a digital delay, and that's how they did the handclaps on "Benny and the Jets," but most of it was tape slap. They used very, very basic things, so I try to stay with very basic stuff. If I do use anything high-tech, I try to make it sound transparent.

Has the Elton stuff influenced how you work on other things?
Yeah, it has. It's influenced me to stay simpler because I think I can get better imagery. I remember hearing Elliot Scheiner saying at a surround conference, "More often than not, if I want something to be in a speaker, I put it there. I don't often try to occupy the space between monitors [phantom image], because I find I get a lot of phasing and strange clarity issues." I've adopted my own

version of that. It depends on the instruments that I'm using, but I have a fairly formalized way of placing things now, and it seems to work. I know where I like to put drums, I know where I like to put the bass guitar, all the way down to background vocals and strings and things like that, although I never start a song with a prepared template. I always start a song from the ground up, and depending on what the song dictates, I'll change the placement.

Do you think you'll still stay in the box for all your projects?
I would like to do some things that are initiated in surround and are repurposed later into stereo, which is sort of the reverse of the way things are done. I'll probably stay in the box for that not only for simplicity, but because that's where the project started, and it will probably organically take shape that way.

I think it would be different if I had a different budget structure to work within, but there just isn't a lot of support financially for mixing in surround, so you have to find a way to be clever if you want to do it, and that makes me stay in the box. Oftentimes now I'll work on something for a half an hour, and if it's not happening I can just pull another track up, so I can work on three or four songs a day. Nothing ever gets boring and nothing ever suffers because you're trying too hard, which I think is really the beauty of it.

Chapter 29

Dave Pensado

Of all the genres of music, mixing R&B may be the toughest, thanks to the almost constant change in what's state of the art and the artists' and producers' penchant to experiment with new sounds. Over the last two decades, Dave Pensado has taken mixing to a new level in artistry, having mixed big hits for superstars such as Christina Aguilera, Justin Timberlake, Kelly Clarkson, Pink, Black Eyed Peas, Beyonce, Shakira, and Michael Jackson, among many others. Well known in the business way before his popular Pensado's Place web series, Dave not only is on the cutting edge of technology, but also has thought long and hard about the more cerebral aspects of mixing. For more about Dave, check out pensadosplace.tv.

What's harder to mix—an R&B or a rock track?

I mix both, and R&B is infinitely harder to mix than rock. Think of it this way: Let's say you're painting a portrait. Rock is like having the person you're painting sitting in front of you, where you look at them and paint, and look at them and paint, so you always have a reference. In R&B, there is no reference. It's like trying to do a portrait from memory, but because you don't have the person there, you can paint something that transcends what he is. You can make him prettier, you can make him uglier, or you can make him abstract if you want.

R&B gives you fewer limitations and a lot more freedom. We don't have to have the snare drum sound a particular way. It can sound like anything from an 808 to a hand clap to a little spitty sound to a rock sound; but you put certain snare sounds in a rock song, and it's just not a rock song anymore.

Do you approach EDM differently, too?

Yes, I do. I would like to think that I approach every mix differently, much less every genre. When I'm changing styles of music, I tend to immerse myself in that type of music for a day beforehand so I can clear my musical palate, so I'll listen to a lot of EDM just to get in that frame of mind. The other thing is that when I'm starting an EDM mix, I like to make sure that I have the newest, hippest plugins around. In that genre sounds move so fast that you have to have the freshest stuff to create separation between you and the latest songs that are out there.

Do you hear the finished product in your head before you start a mix?

Yeah, I really can. I might not have 100 percent of the final product in my mind when I start, but I pretty much have it outlined. Then as I start filling in the outline, sometimes things change a little bit. Every once in awhile, maybe one out of two or three hundred, I might just pull everything down and say, "I don't like any of this," and start again from scratch.

What's your approach to using EQ?

Well, I think of EQ as an effect, much the same way you would add chorus or reverb to a particular instrument or vocal. I tend to be most effective when I do the standard equalizing; then

take it to the next level. For instance, I might take a vocal to where I think it's really EQ'd nicely, but then I might add a little more 3k just to get it to bite a bit
more so it brings out a little bit of passion in his or her voice, just to make me feel like the singer was trying harder.

Are there certain frequencies that you keep on coming back to?
I notice that in a broad sense there are. In other words, I always have to add the frequencies from, say, 120Hz down to 20 cycles. It seems like I'm always having to add the frequencies from about 10k on up as well.

So much of the music that I do is programmed, and that gives the producer the luxury of pretty much getting the sound he wants from the start. In the old days you always pulled a little 400 out of the kick drum, and you always added a little 3k and 6k to the toms. That just doesn't happen as much anymore because when I get the file, even with live bands, the producer has already programmed or triggered the sound he wanted off the live performance, and the drums are a lot closer to being finished. It frees me up because now I have the luxury to really get inside the tracks within the timeframe I'm given, whereas before I would have to spend that time just getting it up to a certain level. Most of the stuff I'm given now is really starting out in a lot better shape than it used to sound-wise.

How about panning?
I think that there are three sacred territories in a mix that if you put something there, you've got to have an incredibly good reason—that's extreme left, center, and extreme right.

I've noticed that some mixers will get stereo tracks from synthesizers and effects, and they just instinctively pan them hard left and hard right. What they end up with is these big train wrecks out on the ends of the stereo spectrum; then they pan their kick, snare, bass, and vocals to the center and there's all this stuff stacked on top of each other. If it were a visual, you wouldn't be able to see the things behind the things in front.

What I do is take a stereo synthesizer track and just toss one side because I don't need it. I'll create my own stereo by either adding a delay or a chorus or a pre-delayed reverb or something like that to give it a stereo image. I'll pan maybe the dry signal to 10:00 and then I'll pan the effects just inside the extreme left side. I would never put it hard left because then there are too many things on top of it. I might pan it at 9:00 and then pan the dry signal to, say, 10:30 or something like that.

Do you use a lot of compression?
I look at compression as having two functions—as an effect, and when you want to keep a particular sound right up front in your face in the mix. I very rarely use a compressor to even out dynamics. Dynamics are something that I just can't get enough of. I once read an interview with a well-known engineer in which he was praising a particular compressor for its ability to take the

dynamics out of a drum performance because the drummer would get happy on the first downbeat of every chorus and play a little louder. I thought, “I spent my whole career trying to add those dynamics and trying to make the drummer sound like he got happy going into the chorus.”

The compressors I like the most tend to be the ones that actually help me get some of those dynamics. That might be a contradictory statement, but if you’re careful with the attack and release times, you can actually get a compressor to help you with it.

How do you do that?
A lot of times what I’ll do is put the effects only on the compressed sound. As a result, the reverb actually has a snap and aggressiveness to it. Every once in a while I’ll make it stereo, where I’ll put anywhere from a 9- to 15-millisecond delay on one of the channels so the tight compressed sound is out on the edges of my stereo spectrum, but the original sound’s in the center. That creates an incredibly nice image, particularly for ballads and slow tunes where you have a lot of space between the downbeats. That setup works great for snares, kicks, and hi-hat. Every once in a while, it’ll make a guitar come alive, too.

What you’re doing is controlling the dynamics, but you’re actually increasing the dynamics, too. It’s the strangest thing because psycho-acoustically it’s not getting louder, but your mind is thinking it is. On the radio, it just jumps out of the speakers.

Do you use much mix buss compression?
I rarely do anything to add color on the master buss. I was taught that putting things on the master buss is an excuse for not doing something earlier in the chain. If you have to add a lot of anything to the master buss, then you probably haven’t worked enough on the individual elements. For me, a dB here or there is okay, but I try to keep things minimal. If I’m in the box, I’ll try to use some [Waves] L2, but if I’m not in a box, I’ll use the SSL board compressor if I use any at all.

How much of your mixing is in the box these days?
About 60 percent. It’s all dictated by the budget. Sometimes even the clients that can afford it would rather I mix in the box to save some money. I think I can get it pretty close to the sound of a console—maybe to within 2 percent or so.

Do you have a philosophy about adding effects?
The way I think of it is the pan knob places you left to right while the effects tend to place you front to rear. That’s a general statement, but it’s a good starting point. In other words, if you want the singer to sound like she’s standing behind the snare drum, leave the snare drum dry and wet down the singer, and it’ll sound like the singer is standing that far behind the snare drum. If you want the singer in front of the snare drum, leave him dry and wet down the snare drum.

That said, I like a vocal mostly dry, but then it usually doesn’t sound big enough. You want the vocalist to sound like they’re really powerful and dynamic and just giving it everything, so I’ll put

an eighth-note delay on the vocal but subtract a 16th-, a 32nd-, or a 64th-note value from that eighth note. What it does is gives a movement to the delay and makes the singer have an urgency that's kind of neat. I put the eighth minus 1/64th on the left side, and put the straight eighth note on the right side. You can experiment with pushing the pitch up a little bit on one side and down on another, too, if your singer's a little pitchy, since that usually makes them sound a bit more in tune.

Sometimes putting the eighth-note triplet on one side and the straight eighth note on the other, if you've got any kind of swing elements on the track, will make the vocal big, yet it doesn't make the singer sound like he's taking a step back.

Another thing I like to do is to take the output of my effects and run them straight into another effect. I'll take an exciter and just dump the output straight to a chorus so it's only chorusing the high frequencies. I think that's more pleasing than having low notes chorusing all over the place. Another thing I'll do is set up a chorus, and I'll pan one side hard left and the right return at 2:00, then I'll take another chorus and pan it hard right, and then the left return from that one I'll pan at 10:00. Now the lefts and rights are kind of overlapping. On one I'll have the chorus depth just a little less than the other, and I'll have the other modulating a third faster. When you add a vocal to that, you get this real nice spectrum that just widens it out because there's an equal amount of chorus, yet one of them is chorusing deeper and slower than the other one. If that's not wide enough for you, add a delay in front of both of them that's different on each side and then add that to your background vocals. They don't take any steps back in the mix, but they just get fat.

A lot of times I'll take two reverbs and instead of running them stereo, I'll run them mono in and mono out and pan one just inside the left and one just inside the right. I'll use the same program on both, but I'll slightly alter the values. Just return it in mono, and you'll be surprised by how much better it sounds sometimes.

What operations do you still do in the box even though you may be mixing on a console?
When I mix on a console, I mix 100 percent differently from when I mix in the box. When you work in the box you have to make some decisions about how you're going to make it sound, especially if you have someone who's used to listening to analog, because you don't have a console to supply that sound. You have to give the producer or artist something that they've been accustomed to for a long time.

Immediately you start thinking in terms of the tape machine emulation plugins (my favorites being the Waves and UAD, although Steven Slate just came out with one that's good). I might take my drums and put them on an aux, then put a tape machine plugin across that aux. Or I might use a plugin by Dave Hill called Phoenix to add some color or use the AC series from McDSP. You have to practice with every one of those saturation plugins to see what they do best because there's not one flavor that will work in every situation.

Are there any go-to plugins that you always use?

There are probably too many to list. [Laughs] I use the P series and sometimes the E6 from McDSP. I use the Waves plugins that everyone knows: the C1 series, the Renaissance Compressor, and TrueVerb. I use the Massey L2207 on just about every mix. I love his TD5 Tape Delay as well. I also use the stock Avid plugins. Those are monsters, and they're very efficient. I'm a big fan of the Air series.

At some point in the mix you have to start choosing plugins based on their efficiency instead of only their sound. There are some that take up so much processing that you can't use them if you're at the end of the mix because you'll run out of processing power. Anything from UAD, Steven Massey, or McDSP works well in that regard because they're all so efficient.

How long does it take you to do a mix these days?
If I had to put a number on it, I'd say that hip-hop takes about 10 hours; pop about 14 hours; rock that's recorded well about 10 hours—and if it's not recorded well, then maybe 16 hours; and EDM anywhere from 12 to 18 hours depending upon the subgenre.

The amount of time is directly proportional to the quality and the number of tracks that I'm given. If doing a mix with four different lead singers, that's going to take me a lot longer than doing one with a single rapper or an instrumental EDM song. Another thing that slows me down is when the session doesn't sound like the producer's vision. In other words, when I pull the session up it sounds like one thing, but when the producer comes in he has a completely different idea of how it's supposed to sound. Those take a long time and add maybe about 50 percent more time to the mix.

What's the average number of tracks you're given?
I never really count the tracks, but when I first pull up a session I have to make a decision on how many voices I want the DAW to use. I'd say it's rarely ever fewer than 90 and usually averages between 120 and 160. The song I'm mixing now I'm struggling with because I only have 192 voices on my rig, and that's not enough!

Is this because they expect you to figure out what to use from all the tracks?
In the world that I work in, one of the major changes over the years has been the removal of any need for commitment. Back when I was working for Babyface, he would record six to eight reels of background vocals [which might have up to 20 tracks of vocals on each reel], which is a lot of tracks, but I never saw them because he bounced them all down to two or three stereo tracks for me to mix. Nowadays I never see anything like that, and I haven't seen it for at least five years. I get every single track that's been individually recorded. Nobody wants to commit anymore. That's okay, I don't mind doing it. It gives me a little more flexibility, and we usually end up within 5 percent of what the producer wanted anyway.

Another change from the old days is that the rough mixes should probably be called "reference mixes" because there's so much time spent on them. Sometimes they'll spend two or three weeks

on one and even bring in an engineer specifically for that process. They're not rough mixes at all; they're pretty much what the client wants. I think it has made us all better mixers, because 10 years ago the gap between what they gave me and what they got back was a lot wider. People would walk into my control room and say, "Holy crap! That's incredible!" You don't get as many of those moments anymore, because they've spent so much time getting it so close before I even get it. The good news is that allows me to spend more time on the small things to make them a lot better.

The other big change is that five years ago, if you didn't like something, the tiniest change would end up costing at least five grand, which isn't the case now because you never really finish a mix that's in a DAW. I get calls on mixes that I completed six months ago, asking me to change a word in a chorus. I don't think that's a bad thing, but it all goes back to the lack of commitment. I think creativity works best when there's a deadline. We're never in a finished state now, because everyone has a DAW at home and knows how quick it is to pull something up and change it.

What alternatives mixes do you deliver?
Ninety-five percent of the time they just want the main mix. If they want the vocal up, they just call, and 10 minutes later they have a mix with the vocal up. In terms of the classic stems of a cappella, TV mix, and instrumental, I don't even get many requests for that anymore unless the artist is performing live. Sometimes they ask me to send the entire session over to the live mixer, which I don't understand because I'm using plugins that not many of those guys have.

What monitors are you using these days?
I hate to bore you, but it's still NS10s and Auratones. If I'm working at Larrabee [Dave's Hollywood-based commercial studio of choice], I've got the Augsburgers [main monitors] and all the rental monitors available if someone wants to hear it like that. [Producer] Ron Fair gave me a pair of KRKs with a KRK sub so he would have that comfort factor when we work together. I encourage that from all the producers I work with. [Laughs]

The NS10s give me all the low end I need. When you see the specs on a speaker like an NS10 that says it's down 20dB at 80Hz, that doesn't mean that you can't EQ 80Hz, because it's really there; you just have to train your ear to hear it at 20dB down and not flat. To get your ear trained just takes a while until you do some mixes and hear what they sound like in a number of places, then determine what the strengths and weakness of that speaker are, and then mix accordingly.

Do you have any listening tricks?
Yeah. I have my sub connected so I can monitor it without hearing the mains to know where my low end stands. I've done it so much that I can get the low end right just by listening to it that way. Then sometimes I walk into the hallway to listen to a mix, and that helps me get some balances, and the old car test has always been a great tool that we all use. Now more than ever, it's integral to the process if I'm working in someone's private studio.

What level do you usually listen at?
I usually listen to NS10s at kind of a medium level, and Auratones I listen to at the same volume you would listen on a TV. I found that in order for the NS10s to really work, it's best to have them stay at one level for most of the mix. Near the end of the mix, I'll check the EQ with the NS10s about 20 percent lower and again about 20 percent higher.

When I'm at Larrabee, I'll use the big speakers mostly to show off to clients and to just have fun. I like to turn it up, and if my body is vibrating properly, then I'm happy with the low end. A lot of engineers use them to hype the client, but I also use them to hype myself! If I'm cranking and I'm not getting excited, then I just keep on working.

What I recommend is to train yourself to work at three volumes and mark them so you work consistently at them. After a while you'll do it without even thinking. That's what you want from the entire mix process. You want to be removed from any technical thoughts and be 100 percent in creative mode.

What advice do you have for someone who wants to get better at mixing?
Keep an open mind to all music. You'd be surprised by what you can learn from records outside of the genre that you normally work in. Focus on things that can expand your life, and they will expand your creativity as well. Life experiences are always going to contribute to your creative process.

When it comes down to it, emotion and passion are at least as important as technique. The way I can prove that's accurate is the number of times a rough mix gets used over the mix of a top-level engineer and goes on to sell millions of records. Creativity is about the spur of the moment and not thinking about the process. Emotion always trumps technique and technology.

Chapter 30

Elliot Scheiner

With his work having achieved tremendous commercial success, Elliot Scheiner has also attained something far more elusive in the music business—the unanimous respect of his peers. Indeed, if you want a mix that's not only a work of art, but also a piece of soul that exactly translates the artist's intentions, then Elliot's your man. With a shelf full of industry awards (seven Grammys, an Emmy, four Surround Music awards, the Surround Pioneer and Tech Awards Hall of Fame, and too many total award nominations to count) from his work with the Eagles, Steely Dan, Fleetwood Mac, Sting, John Fogerty, Van Morrison, Toto, Queen, Faith Hill, Lenny Kravitz, Natalie Cole, the Doobie Brothers, Aerosmith, Phil Collins, Aretha Franklin, Barbra Streisand, and many, many others, Elliot has long been recognized for his pristine mixes.

Do you have a philosophy about mixing?

I've always believed that if someone has recorded all this information, then they want it to be heard. Back when I started, if you were able to hear every single instrument in a mix, that was considered a huge achievement. Granted, there wasn't as much information when I started as there is now. I've come across sessions that are a hundred and some odd tracks wide, so it's not as easy for everything to be heard in a track like that.

I have to admit that the way some people record things today is a bit peculiar. All of a sudden you'll be dealing with seven or eight different mics on the same instrument. It's mind boggling that you have to listen to every single channel to decide which one you want to use. If you pick the wrong ones, they come back at you and say, "Oh, we had a different combination," or "It doesn't sound quite right to us," but they don't tell you what they did in the first place! Granted, mixes can be a little more difficult to deal with because of those issues today, but I still take the same approach of having everything heard with every mix.

If you have a hundred tracks, will you try to have them all heard, or do you go in and do some subtractive mixing?

Well, it depends if that's necessary. I don't usually get those kind of calls where they say, "Here's a hundred tracks. Delete what you want." Usually I'll get between 24 and 48 tracks, and hardly am I ever given the liberty to take some of them out. I just assume that whatever an artist and producer sends me is kind of written in stone. They've recorded it, and unless they tell me otherwise, I usually don't do subtractive mixing.

How often do you work at home?

That's where I mix almost all of the time these days because people don't want to pay to mix in a commercial studio.

You're not mixing in the box at home, are you?

No. I've got a Yamaha DM2000 digital console and a bunch of outboard gear as well.

How long does it take you to do a mix on average?
Depending on how complicated it is, it usually takes anywhere from three hours to a full day.

Three hours is really fast!
Well, a lot of time you just get a vibe and a feel for something, and it comes together, and then you look at it and say, "How much am I actually going to improve this mix?" If it feels great and sounds great, then I'm a little reluctant to beat it into the ground.
For me it's still about a vibe, and if I can get things to sound good and have a vibe, that's all I really care about. I still put [legendary engineer] Al Schmitt on a pedestal. He can do three songs in a day, and they'll be perfect and amazing sounding and have the right vibe, so it's not like it can't be done. Some people say that you can't get a mix in a short time, and that's just not true; Al's my proof.

Where do you usually start your mix from?
Out of force of habit, if there's a rhythm section I'll usually start with the drums and then move to the bass and just work it up. Once the rhythm section is set, I'll move on to everything else and end with vocals.

How much EQ do you use?
I can't say that there are any rules for that. I can't say that I've ever mixed anything that Al has recorded, but if I did I probably wouldn't have to use any on it. With some of the stuff done by some of the younger kids, I get it and go, "What were they listening to when they recorded this?" In those cases I'll have to use drastic amounts where I'll be double compressing and double-EQing—all kinds of stuff in order to get something to sound good. I never used to have to do that. Obviously those mixes are the ones that take a day or more.

Do you have any go-to plug-ins that you use?
The only plug-ins I use are the UAD plug-ins. If I need to fix something, then I'll use some of the iZotope plug-ins, but I generally don't use too many. To this day I've never utilized a reverb inside the box. I just don't like the way they sound, so I'm still going to external devices for reverb. My favorite is the [Lexicon] PCM 96. I'm still using a lot of the reverbs built into the desk. I do use the EQs in Nuendo extensively when I need it. I love their EQ.

When you're setting up a mix, do you always have a certain set of effects, like a couple of reverbs and delays, ready to use, or do you patch it as you go?
Usually I don't start out with any reverbs. I'm not one for processing. I'd like to believe that music can survive without reverbs and without delays and without effects.

Obviously when it's called for I'll use it, but the stuff I do is pretty dry. The '70s were a pretty dry time, and then the '80s effects became overused. There was just tons of reverb on everything.

Most of your Steely Dan stuff you did is pretty dry, isn't it?
It's pretty much dry. If we used anything, we usually used plates.

In the days when I was working at A&R [the now-defunct New York City studio owned by producer Phil Ramone], we had no remotes on any of our plates there. Phil wanted to make changing them difficult because he tuned them himself and he really didn't want anybody to screw with them. There would be at least four plates in every room. Some of them might be a little shorter than another, but generally they were in the 2- to 2 1/2-second area. There was always an analog tape pre-delay, usually at 15 ips, going into the plates. The plates were tuned so brilliantly that it didn't become a noticeable effect. It was just a part of the instrument or part of the music. You could actually have a fair amount on an instrument and you just wouldn't notice it.

How much compression do you use?
Very, very little if I can, but it depends on the project. Growing up at A&R, there were only four pieces of outboard gear in every room: a pair of Fairchild 670 compressors and 2 Pultecs [EQs], and that was it. I used a Fairchild for the bass and the kick drum. There was no EQ on the consoles because we used broadcast consoles, so you were very selective about what you put that outboard gear on. It was usually something where you had trouble getting a sound. It was more about moving the mic or the musician or having them play differently than EQing or compressing.

Most of the compression I did then was hand compression, where you rode a fader and you learned how to ride it smoothly so you didn't screw up. I never got into the habit of relying on compressors for anything. I probably do a little more now than I did in the past. I use a stereo compressor across the buss usually when I'm mixing, but it's not for compression as much as it is for the sound of the compressor.

What do you use?
I use a Neve 33609. There's very little of it going on; the needle barely moves, but I like what it does to the sound. Sometimes I'll put a GML EQ across it as well just to add a little air on the very top.

What do you use for monitors and how loud do you listen?
I'm still using NS10s for stereo and five of them for surround. Generally when I'm mixing, I monitor at about 40dB.

Wow, that's quiet.
Yeah, I just find that I can definitely hear the balances better at a lower level so I can hear if I've got a little too much or not enough of something. That comes from wanting to hear every instrument. If I can hear it when it's soft, then it's probably going to be there at any level. I'd like to believe that I can still hear a little bit now because I've monitored so softly for so long.

Do you go up on bigger speakers to get your low end?

No, I might go up to 80 or 85dB when I'm putting the kick and the bass in just to hear where they're sitting comparatively, but once I establish that, I'll go right down to 40 or so.

Do you have any listening tricks, such as going outside the control room or anything like that?
I used to take a mix home to listen on my speakers there, but then I found out that I was getting better results by listening in whatever car I was driving. A lot of guys used to base their mixes on what it sounded like in a car because a lot of the times the car was the lowest common denominator.

How many mixes do you deliver?
Usually just two or three. Pretty much vocal up and vocal down unless there's something so weird that I'm worried about it; then I might do an alternate version with something lowered or out. I've got to say that with most engineers, your initial instinct is usually correct.

Do you draw your automation in on the workstation or do you only use fader automation?
I use the console automation as my primary automation, but occasionally I get more tracks than I have outputs on the DAW so I'll combine tracks inside the box and use the automation in Nuendo to do whatever needs to be done.

Do you have any special tricks that you use when you're mixing?
I've never relied on any trick stuff. I wish I could say that I've been using something all these years that's some kind of secret, but I don't do anything that's of a secret nature. I'm more than willing to tell anyone who asks me what I did because I've never had any tricks. I've always felt that the real trick is in how you hear it.

Let's talk about surround, since you've done so many surround mixes. Has your approach changed over the years?
In some regards, yes, but as far as the overall approach, no. I still do a very aggressive mix. I like as much information coming out of the surrounds as I do the front, so I'm still as aggressive as I was when I first started, maybe even more so. The only thing that's really changed for me is how I use the center speaker. I try to use it a little more than I have in the past. I've found that when listening in a car, the center speaker is a little more important than it is in the home. If I put a vocal in there, it's going to be at least as loud as the phantom center, maybe a little more.

Do you only put the lead vocal in the center?
No, I put the snare, bass, kick drum, and sometimes an instrument that's a mono track, like a tambourine or bells or something that I can just put in one speaker.

Do you still keep the mix fairly dry?
Yeah, pretty much.

Do you use the LFE much?
I run the speakers full range so I don't have to put all that much in the LFE. I always worry about bass management in people's homes, since every receiver does it differently and there's no way to predict what will happen, so I just ignore it and put a minimum of stuff into the LFE.

What's your approach to doing a live concert as compared to a studio record in surround?
I'm probably the only guy who does it, but I don't believe in setting an arena perspective in surround. I don't like the idea of having just some ambience and room mics in the back. I still want to hear the music coming from everywhere. I still try to be as aggressive with live stuff as with a studio record, so I take the creative license to move to the stage and put instruments in the rear.

I think if the rear speakers aren't lit up, the listener feels cheated.
Yeah, I agree with you. What really gets me off is hearing information coming from the rears because that's what's unusual about surround. We've been listening to the front side for as long as we've had recordings. Now that we have other speakers, that's where a good deal of the information should be coming from.

Would you have any advice for someone who's just starting to mix?
I would say that you have to believe in yourself. You can't second-guess what you're doing. I've always been of the mind that if I can make myself happy listening to a mix, then hopefully the people who are employing me will be just as happy.

I don't try to guess what someone might want. If there's someone there in the room with me when I start a mix, I know that sooner or later I'm going to hear whether they hate it or they love it, but generally I try to mix for myself. At this point in my career I know that if people are calling me, then they must like what I do. Just remember that what we do is to convey the artist's feelings and make it as musical as possible without harming it.

Chapter 31

Andrew Scheps

Andrew Scheps has worked on mega-hit albums for a who's-who of superstar artists such as Red Hot Chili Peppers, Metallica, U2, Justin Timberlake, Jay-Z, the Rolling Stones, Linkin Park, Jewel, Neil Diamond, and Adele.

Even though he's working out of his pretty outstanding home studio built around dual Neve 8068s, a massive wall of outboard gear, and dual Studer A800 24-track tape machines, amazingly Andrew is not one still living in the analog past, as the DAW is an integral part of his workflow. To find out more about Andrew and to see pictures of his great studio, go to punkerpad.com.

Can you hear the final product in your head before you mix?

If I know the song, then I already have a pretty clear picture of what I'd like it to be. If not, I'll usually get that the first time I listen through a track. It's not so much for the sonics, but more in terms of size, like figuring out how big the chorus will be. Sometimes I'll get really specific ideas about effects that I'll try as well.

In terms of starting a mix, I think the main thing, especially if it's a song I haven't recorded, is that I go through instrument by instrument to see how it sounds, but what I'm really doing is learning every single part so that when I come to build my balance, I know where everything is going to be.

Do you have a template for your effects before you start to mix?

Kind of, although I don't use a lot of effects. I use a lot of parallel compression so that's more of what I have set up. In terms of what gets sent to those compressors, some of it is consistent and some of it changes with every mix, but they're ready for me at the push of a button, which on an analog console is great because I just leave that part of the patchbay alone.

In terms of effects, sometimes I'll have one kind of chorus-spreader kind of thing and one reverb and that's it. I don't tend to use many effects because a lot of the stuff I mix is straight-up guitar rock, and it's more about the balance and making things explode.

Do have an approach to doing that?

You're never really as aware of your own process as you think you are. I'll think that I really didn't do much of anything, and then I'll look at a mix and find that I'm using 50 things on it.

Also, because I mix on a console there's the whole process of laying out the outputs of Pro Tools to see where everything is going to come up on the console. There are things that always live in the same place, like channel 24 is always the vocal, so I'm usually figuring out how to lay out everything between the drums and the vocal. I do that while I'm finding out what everything is doing, so there's a long discovery process where it doesn't seem like I'm getting much done, but

then everything happens really quickly after that.

Where do you build your mix from?
It depends. I'd love to say that I always build it from the vocal, but usually what I'll do is deal with the drums to get them to act like one fader's worth of stuff instead of 20 or whatever it is. Once I've gone through that process that I just described, everything seems to come up at once. I'll have listened to vocal and the background vocals and know exactly where they are, but I'll get the band to work without the vocals first, which I know a lot of people don't think is a good idea.

I think it's the same thing when you're working on a particular instrument in solo. After 20 years, my brain sometimes unconsciously knows what an instrument will sound like soloed, so I'll tend to get the tone on things separately, and then it's all about the balance. I almost never have to go back and change things once I get the vocals in. My brain seems to know what that balance is going be when the vocals are inserted.

How much do you do in the box?
I always think that I do nothing in the box, but I really do a lot of the technical things. The EQs on the Neve are very broad and very musical; they're not good for anything surgical. If there's a nasty frequency in the overheads or the snare is ringing too much, I take care of all of that in Pro Tools. Usually I'll have the background vocals coming out of one stereo output pair, so I'll deal with them in the box. Sometimes I might split a couple of them out, but I don't want 20 tracks of background vocals on the console; it's just a waste. A lot of the crazier effects can come from plug- ins there as well.

There's quite a bit that goes on in Pro Tools, but it's more about shaping things before they get out into the console. The console is much more of an organic balance thing, while Pro Tools is more for making things sound the way I want them to sound. The console is more about putting it all back together and mixing it.

I actually mixed in the box for years in this same room. I had a [Digidesign] ProControl in here, and that was great. In fact, there are some things that I mixed in the box that I listen to now and go, "Wow, that sounds really good."
I don't have any philosophical differences with mixing one way or the other way. It's more of once you have the console, as much of a drag as it is to document everything, it's such a joy to mix on it. When I'm mixing, it doesn't matter whether it's coming off tape or Pro Tools; it's just faders and speakers, and that's it. I love that because sometimes mixing in the box makes you so precise that you then fix things that don't really need fixing. I like the sloppiness of doing it on the console.

Do you find that you're using your outboard gear less?
No, not at all. When I document every mix, I wish that was the case because it's a lot more to write down, but because a lot of it is parallel processing and stays patched in, it's so much faster

for me to hit a button on the console than it is for me to set the same thing up in Pro Tools. I may send the bass, the guitars, and the background vocals to a stereo compressor, and in doing that in the box, it could change the balance on the board, so that doesn't really work for me at all. It's less of a sonic thing than a convenience thing.

What do you use the parallel processing on?
On this mix right now there's a parallel compressor on the kick and snare, then there's another just on the snare. There's a stereo one on the toms and overheads, a mono one on just the dirty bass (this song has three basses), a stereo one on the guitar and vocals, and then a couple of different ones just for the lead vocal—one that's sort of spitty and grainy and one that's sort of fat. That changes from mix to mix. In fact, it changes a lot.

Are you tucking the parallel processed channel just underneath the unaffected one?
Yeah, although sometimes the parallel one ends up being pretty loud, in which case it's almost like using an insert compressor, but it's across a few things. Sometimes it's just tucking it in to add power or weight.

Do you EQ the parallel processing as well?
Not much, because everything is post-fader, so it's only the EQ'd stuff that gets in there. Sometimes on the drums that will start to really bring out a badly ringing cymbal, so I'll go in and do something surgical to fix the problem, but most of the time it's the same tone as the uncompressed.

Are you using buss compression as well?
Yeah. I used to never compress the mix because I could never find one that I liked that didn't take away from everything else that was going on. Then a couple of years ago I started using the 2264s [the onboard Neve compressor modules] that are in the console. I also learned the lesson that if you're going to compress the mix, you begin your mix with the compressor already on. You don't get your mix and then put your compressor on, because that doesn't work. You have to mix to it.

I don't use heavy compression, though. I don't think I ever add more than 3 or 4dB, so I'm not really smashing it. It is a pretty aggressive setting, though, like a super-fast release. I've tried printing uncompressed mixes and bringing those to mastering, but you can never re-create the sound, so I always mix to it now.

How many alternate versions of mixes do you do?
If there's nothing specific that comes up during the mix, I'll do a full mix, an instrumental, sometimes a TV mix just because people are used to getting it, a vocal up and a vocal down, and an a cappella. If there's been a lot of talk about the balance of the backgrounds, I'll also give them an a cappella lead and an a cappella background. If there's been talk about any particular instrument that the band doesn't agree about, then I'll print them an alternate of the other variation they're considering and then strongly label the one that we think is the right one.

How long does it take you to do a mix?
It really depends upon the material. If it's well recorded and I've already done a song for the album, then it can come together in as little as three or four hours. The first one on an album usually takes longer because there's a lot to sort out, so it will usually take a full day. I'm usually ready to play something for the band by late afternoon at the latest.

What are you using for monitors?
I love the old Tannoy SRM-10Bs. One of the first people I worked extensively with had these speakers; then I borrowed a pair for a gig and just fell in love with them. Since then I've tried seven or eight times to find something else to use beside them, but their midrange and top end are different from most other speakers, which makes it really hard to go back and forth. Now that I work mostly in my own room, I've gotten to the point where I don't have to listen on anything else. I never switch speakers and I never listen in the car when I'm mixing, yet almost never does anyone say, "Ah, there's a problem with the low end." At this point I own three pairs here and another pair in Europe, and that's all I'll ever use because I know that I can walk into any room and be safe with what I'm hearing. I power them with an old Crown DC-300A, and they just match well. Whenever I take the speakers somewhere else, though, I just use the amp they've got, and it's usually fine.

How loud do you monitor?
When I'm getting the mix together, I monitor pretty loud and for longer than I probably should. Once I get the balance, then I mix really quietly, but still occasionally check things loud every once in a while. Mixing is such an emotional thing. You're trying to get it to seem exciting, especially on the rock stuff, so you have to hear it loud to know that the kick and the snare and vocal are hitting you in the chest.

There are things that you can't judge when it's loud, though. You can't judge the vocal level properly because the vocal will sink into the mix more when it's loud, but in terms of impact and emotion, you've got to crank it.

You said that you mixed in the box for a long time. How did you get back into using a console?
For *Stadium Arcadium* [by Red Hot Chili Peppers], the band wanted the record mixed on a Neve. It was tracked on a Neve, and they wanted it mixed on one, too. I tried all of the available Neve rooms in town, and it just wasn't working for whatever reason, so I ended up renting a console and realized that this room could easily accommodate it. The other thing is that I just loved having a console, and as an investment, it's not going down in value—unlike an SSL, whose values are still plummeting.

The first desk had 32 inputs, but when I knew that I was going to be mixing Metallica, I knew that I needed more. Thirty-two inputs plus the other 10 on the BCM-10 [the small Neve broadcast

desk that Andrew uses as a sidecar console] wasn't going to cut it, since their drums alone had around 30 channels. They weren't all going to be coming up on the board, but 32 just wasn't enough inputs. I was going to rent the console that I used for *Stadium Arcadium* again, but in the process convinced them to sell it to me. As far as the outboard gear, studios keep closing, and that's where most of that comes from, since it tends to come in batches.

Where do you do your automation—in the box or on the desk?

Most of the automation is done on the desk. The only thing done in the box is for extreme fixes. Once the mix is pretty much done and we're adjusting something like background levels with the band, it's easier to do that in the box because sometimes it's just a certain word that they want louder, and then you don't want to be sloppy with a fader ride on the console. It's more precise in the box.

Do you ride the rhythm section for fills or is your mix fairly static?

I still do ride things because the compressors are sucking out some of the natural dynamics of how the instruments were played, especially on some of the louder rock stuff. I'm only adding it back in, rather than creating something out of nothing. I don't turn the whole mix up at every chorus, for instance. Some people can do that with success, but I always hear it when I do it. I do definitely push the drums for the downbeat of the chorus and really try to accentuate anything that might be cool in a guitar performance, as well as some of the idiosyncrasies. The rides aren't drastic, but most things are moving.

Chapter 32

Ken Scott

Legendary producer Ken Scott began his career at the Abbey Road Studios working with the Beatles on *The White Album* and *Magical Mystery Tour*; on six David Bowie albums, including the seminal Ziggy Stardust album; and with Pink Floyd, Elton John, Duran Duran, Jeff Beck, Supertramp, Procol Harum, Devo, Kansas, Mahavishnu Orchestra, and many more. To put it mildly, he's an absolute icon in the recording industry, having been a part of records that have conservatively sold more than 200 million units.

I got to know Ken while co-writing his memoir, *Abbey Road to Ziggy Stardust*, which is filled with stories about him working with some of the greatest artists in the world, as well as a lot of technical bits that engineers love. During the writing of the book, I was scheduled to produce the third album for the band SNEW, and I sheepishly asked Ken if he would track the basics for me. Much to my and the band's delight he agreed, which was the beginning of a great experience watching a man of such legendary status work.

Ken was a study in quality via fundamentals and minimalism. While many engineers wheel in a number of tall racks filled with pricey and exotic gear, Ken used nothing more than what the studio had. During tracking he occasionally used a compressor on the bass, guitar, or vocal, but that was all.

The sounds he got were almost instantly wonderful. A little EQ here and there (and I mean a little), and we were all blown away. It reminded me of how important fundamentals really are and how easy it is to get away from them.

When it came time to mix, I had some pretty good roughs that everyone was happy with, but we all wanted to work with Ken again. Once again, much to our surprise and pleasure, he agreed to do it, and once again we got to see and hear just how effective great technique and ears can be. With the help of only a single reverb, a couple of delays, and sometimes a chorus/stereo spreader, Ken did his thing. Occasionally he used a compressor on one of the instruments and just a dB or so of an SSL compressor across the mix buss, and every now and again a plugin in the DAW. I came away thinking, "So this is how he got the sound on those Bowie records." Big ears and experience trumps tons of gear any day, at least for some kinds of music.

I've learned so much from all the mixers featured in this book, but Ken's lessons are the most profound. A mix can sound great without a huge amount of tinkering if your tracks are great to begin with (which they were because he tracked them). Of course, the ears and experience of working on albums that we all hold as legendary helps as well.

While a lot about how Ken works can be found in *Abbey Road to Ziggy Stardust*, I asked him some mixing questions that we deemed too technical for that book but perfect for this one. To read

Ken's vast discography, visit ken-scott.com. To discover more about the *Abbey Road To Ziggy Stardust* book, visit abbeytoziggy.com.

What's your philosophy about mixing?
It all starts from recording. That's why I don't like mixing other people's stuff—because I find that I can't do as much with it. If I recorded it, there's an inherent thing of everything finding its own place.

I like everything to be heard—not necessarily the first time you hear it, though. If you listen to something a second or third time, you might go, "Wow, I never heard that guitar part back there," but everything has to have its own place. That's probably a question of the EQ and general sounds I use, but I also like a mix to have depth, too, which I do by using distant mics when I'm recording and/or reverb.

The other thing is to make decisions as much as you can along the way. Everyone is too willing to let any decision wait until the last minute instead of making it on the spot. When I started, we had to make the decision on the balance of the instruments or how much reverb we added because we only had a limited number of tracks, but that was good training for later even when we had a lot more tracks to play with. Today you see people piling on track after track and waiting until the mix to sort it out. That makes everything take a lot longer and makes it harder to mix because you're not sure how everything is going to fit together.

I've noticed that there are certain frequencies that you always use. How did that come about?
Because of what I was trained on. The EQ on the EMI desks [back at EMI Studios, where Ken started] was so limited that you tended to get used to certain frequencies. There was 100Hz and 5k on the desk, and then we had an outboard EQ that could give us either 2.7k, 3.5k, or 10k. I usually now EQ somewhere between 4 and 5k, 10k, and more like 60 or 80 these days. Times have changed in that we go for more low end now than we did back then.

The one thing that did change for me over time was my not liking 200Hz. That frequency couldn't be touched in the early days because we didn't have an EQ that was centered there, and it wasn't until later on that I decided that I didn't like it and began to pull it out. I don't know exactly when that happened though, but I know that I was doing it by the time I mixed [Supertramp's] *Crime of the Century*.

Do you pull 200Hz out on everything?
No, bass drum definitely and sometimes the bass.

I know that records from the '60s and '70s always seemed to be centered around the bass instead of the drums. When did the low end on records begin to change?
Probably with the advent of hi-fi systems at home, and when we finally got the option to EQ at a

lower frequency. Once I got to Trident, I started to EQ the lower frequencies because I could, and with the Lockwoods [speaker cabinets with Tannoy speakers in them], we could then both hear it and feel it.

Can you hear the final mix in your head before you start?
No, not the final mix. The general mix, yeah, but quite often I try to get it as close to what I think the final mix is going to be during recording. Stanley Clarke even mentioned about how I had what sounded like the final mix going the entire time we were tracking.

It all comes from the idea that if you don't have the mix going as soon as possible, how do you know what sounds work? You don't know if a frequency on a keyboard or guitar will work together or if something will mask it. You can never gauge it. That's why people take so long in mixing these days, because they don't make the decision up front what it's going to be like, so they never have the correct mix.

You're pretty minimal with effects.
Once again, that comes from the old days, because I didn't have much to use. What I've found is that the more reverbs you use, the more it tends to pull your attention away from the important things in the song because it's not too natural. If there are many instruments playing in a room, everything naturally has the same ambient sound. If you put a different reverb on each instrument, it starts to sound totally unnatural. If you think about all of the Bowie and Supertramp stuff that I did, I used a maximum of two reverbs, and most of it was only one.

When I used two, it would've been an EMT plate and a Cooper Time Cube (see Figure 32.1), so it was one long and the other short.

Figure 32.1: A long out-of-production Cooper Time Cube
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How has your workflow changed?
Number one is that I had to change to mixing a song from beginning to end instead of mixing it in sections like I used to. I'd love to go back to doing it one bit at a time. When I'm doing a short section, I can just listen through to all the instruments one at a time and focus on them better. When I'm doing the complete song I might start listening to the bass; then before you know it, I'll be listening to a guitar instead. Back in the early days before automation, it was like an orchestra

mixing because everyone in the room had things to do, and we all had to do them in sync to be able to get the mix happening. Even though I was doing it in sections, within those sections there were things changing, so we needed at least five sets of hands on the board to get it done.

Were the mixes better because of the "all hands on deck" feel?
Yes, they were more organic. Today, the fact that we can get everything as precise as possible leads us to think about the mix too much. Perfection comes from the soul and not from the brain. Because it's all in the computer you think about it more and don't feel it as much. When you get all of those people working together it feels better, although you can't do the same thing on a computer. The same thing with using a lot of reverbs; it becomes less organic.

One of the things I do like to do is get the mix as close as I can with the board automation, then just go in and adjust it a bit in the box. That's very useful.

You're pretty mild when it comes to using compression, even on individual tracks.
Sometimes I do it more than others. It depends upon the desired effect. I tend to limit heavier on acoustic guitars and sometimes the piano, but it depends upon the part and what is required.

Did someone teach you how to mix?
To a certain degree, yes. That consisted of just sitting there and listening to what Norman Smith, Malcolm Addey, and Peter Bown [the EMI staff engineers when Ken started there] did, and then attempting to follow in their footsteps. It's also getting that confidence that what you're hearing in the control room is correct for you. It's all about the confidence in the monitors for me. If I'm doing a lot of work in a new place, I'll take the first part of the first session and just listen to stuff to get an idea of what it's like.

I noticed that you don't listen on the small speakers much.
I like to work mostly on the big speakers, but I always check things small. I used to check on Auratones, but these days it will be NS10s or something like that. When I work with people who bring their own set of speakers, it just gets more and more confusing.

Every speaker in every room is going to sound different. I can use almost any studio as long as the monitors are good, because then at least you have the confidence in what you're hearing. It doesn't matter whether you're using crappy mics; if you're getting a sound that you like coming off the monitors, then you know it works. If the monitors are off, you have no idea about anything. You can use the most expensive gear in the world, and it can still sound bad.

You listen pretty loudly, don't you?
Do I? [Laughs] When you're working every day, you tend to start to turn it up louder and louder. I know that at the moment I don't monitor as loud as I used to, but it's because I've spent more time working on my book than being in the studio lately, so my hearing's become more "normal." If you're in there day after day, project after project, you tend to want it louder and louder.

Did you ever bring gear with you to the studio?
No, the only things that I ever brought were my guitars, but never anything else.

What would you suggest to become a better mixer?
First of all, always believe that you can get better. The hardest time that I ever had was when I thought I'd achieved perfection with *Crime of the Century*. It wasn't until I started to find fault with it that I could move on. You should always be learning from everything that you do.

These days I can't necessarily say that it's me getting better, but I'm always striving for what I'm working on to be better than I've done before. Plus, you have to be accepting of the situation under which you're working. If you don't always have a guitar virtuoso to work with, don't expect to get that kind of sound or performance out of the guitarist you're working with. You can expect to get the best out of the player and have it add to the overall sound, though.

Is there one instrument that you take more care with than anything else?
Whatever I determine to be the most important thing of the recording, that's what I will spend the most time on or concentrate on. There are some things that I can't stand the mix of today, but it was exactly what we wanted to do at the time and was agreed to by all concerned. It's purely dependent upon the finished product every time.

Chapter 33

Ed Seay

Getting his starting in Atlanta in the 70s engineering and producing hits for Paul Davis, Peabo Bryson, and Melissa Manchester, Ed Seay has become one of the most respected engineers in Nashville since moving there in 1984. With hit-making clients such as Blake Shelton, Lee Brice, Martina McBride, Ricky Skaggs, Dolly Parton, Pam Tillis, Highway 101, Collin Raye, and a host of others, Ed has led the charge in changing the recording approach in Nashville. In this updated interview, Ed also describes how both he and Nashville have embraced mixing "in the box."

Do you hear the final product in your head before you begin to mix?

To some extent I can. Rather than just randomly pushing up faders and saying, "Well, a little of this EQ or effect might be nice," I like to have a vision as far as where we're going and what's the perspective. Definitely, I try to grasp that early on.

Is there a difference between mixing country music and other genres?

Country music is definitely lyric driven. One of the mistakes that some people make when they try to work on the stuff is they tend to downplay the lyric or downplay the lead vocal. In pop and in rock, sometimes you don't always hear every word, and it's kind of okay if it's buried just a little bit, but country is usually not that way. People definitely sing along with country songs, so that's the biggest thing. The vocal rules, but at the same time, it's pretty boring if it's all vocals and it sounds like a country record from the '60s, where you don't have any power in there. There's an art to keeping the vocal on top without making it dominate.

How much do you mix in the box?

I mix in the box about 95 percent of the time. If someone says, "I wanna mix on a big console," I say, "Great. Let's go," but every time I do that I look back and think, "That sure is easier to do in the box." Actually I started mixing in the box back in 1999, so I was one of the pioneers, at least in Nashville. I saw what was coming and embraced it not only as the future, but also as the present. In fact, I mixed the first in-the-box country record to go number one: "Austin" by Blake Shelton [in 2001].

Part of what's really changed in the business recently is that we live in a recall world now. It's so important to have the ability to recall something 10 times to turn something up or down, and that gets cost-prohibitive on a console since your paying for the room and the assistant and everything that comes with a studio. Mixing in the box is really the way records are made these days, and so many of the big records are done that way. I'm glad that I embraced it early.

When you start to mix, how do you build it?

Well, I'll usually go through and push up instruments to see if there are any trouble spots. All this is dependent upon whether it's something that I've recorded or if I'm hearing it fresh and have no idea what it is. If that's the case, then what I'll do is kind of rough mix it out real quick. I'll push it

up and see where it's going before I start diving in.

If it's something that I know, then I'll go through and mold the sounds in a minor way to fit the modern profile that it needs to be in. In other words, if it has a real flabby, dull kick drum, it doesn't matter what the vision is; this kick drum's never going to get there, and I'll do whatever I have to do to make it so. I'll work through my mix like that and try to get everything up into the acceptable range, or the exceptional range, or at least somewhere that can be worked with. It takes a couple of hours to get good sounds on everything, and then another couple of hours to get real good balances. After that I'll do some frequency juggling so that everybody is out of everybody else's way.

The last stage of the mix and the toughest part is the several hours it takes me to make it sound emotional and urgent and exciting so that it's just not a song, it's a record. It's taking it beyond sounding just good and making it sound like an event.

How do you go about doing that?

I try to find what's important in the mix. I try to find out if the lead vocal is incredibly passionate and then make sure that the spotlight shines on it. Or if the acoustics are sitting there but not really driving the song like they need to, sometimes playing with compression on them can make it sound like, "Boy, this guy was into it." Maybe it's pushing and pulling different instruments. Somebody's got to be back, and sometimes it's better when things are back and other things are farther up front. Sometimes it means making sure your cymbals or your room mics are where you can actually feel the guy, or sometimes adding compression can be the answer to making the thing come alive. Sometimes hearing the singer breathe like on the old Steve Miller records. With a little of that, you might say, "Man, he's working. I believe it." It's a little subconscious thing, but sometimes that can help. It's just basically playing with it and trying to put into it that indefinable thing that makes it exciting.

When you're building your mix, are you starting with bass first or starting with the kick drum?

I start with the kick drum sound, but then I put up the drum kit and put the bass in. Then I'll push up all the static channels that aren't going to have giant moves, like the acoustic stuff, keyboard pads, maybe a synth or Rhodes or piano that doesn't have a whole bunch of stepping-out licks.

Early on, I'll try to make sure that there's room for the lead vocal. I think one of the big mistakes is to work on your track for eight hours and get it blistering hot and barking, but then have no way for this vocal to cut through. You're then faced with the choice of turning this baritone vocal into steel wool with ridiculous EQ, or just turning him up so loud that he sounds inappropriate. It's cool to have a bright record as long as everything kind of comes up together, but if you've got an incredibly bright snare drum and the vocal's not so bright, then it makes the vocal sound even duller. If you're thinking all the way to the end when you master the record and add EQ, it'll brighten the vocal, but it's also going to bring up the snare even more, so you have to have

everything in perspective.

Eventually I get the vocals in and get the backgrounds around them; then I put up the solos and the signature stuff. At that point I get an overall rough balance of everything that sits there pretty well and then juggle the pieces.

What's your approach to EQ?

I just try to get stuff to sound natural, but at the same time be very vivid. I break it down into roughly three areas: mids, the top, and the bottom; then there's low mids and high mids. Generally, except for a very few instruments or a few microphones, cutting flat doesn't sound good to most people's ears, so I'll say, "Well, if this is a state-of-the-art preamp and a great mic and it doesn't sound that great to me, why?" Well, the midrange is not quite vivid enough. Okay, we'll look at the 3k, 4k range, maybe 2500. Why don't we make it kind of come to life like a shot of cappuccino and open it up a little bit?

Then maybe I'm not hearing the air around things, so let's go up to 10k or 15k and just bump it up a little bit and see if we can kind of perk it up. Now all that sounds good, but our bottom is kind of undefined. We don't have any meat down there. Well, let's sweep through and see what helps the low end. Sometimes, depending on different instruments, a hundred cycles can do wonders for some instruments. Sometimes you need to dip out at 400 cycles, because that's the area that sometimes just clouds up and takes the clarity away, but a lot of times adding a little 400 can fatten things up.

On a vocal sometimes I think, "Does this vocal need a diet plan? Does he need to lose some flab down there?" Sometimes we need some weight on this guy, so let's add some 300 cycles and make him sound a little more important. It's kind of contouring.

Also, frequency juggling is important. One of the biggest compliments people give me is that they say, "You know, Ed, on your mixes, I can hear everything." There are two reasons for that. One is I've pushed things up at the right time when they want to hear it, but the other thing is I don't EQ everything in the same place. You don't EQ 3k on the vocal and the guitar and the bass and the synth and the piano, because then you have such a buildup there that you have a frequency war going on. Sometimes you can say, "Well, the piano doesn't need 3k, so let's go lower or let's go higher." Or, "This vocal will pop through if we shine the light not in his nose, but maybe toward his forehead." In so doing, you can make things audible, and everybody can get some camera time.

Do you have a specific approach to panning?

Yeah, I do. The most significant approach is I pan as if I were sitting in the audience, especially with the drums. The reason is, I don't play the drums; therefore, I sit in the audience and listen, and that means with most drummers their hi-hat is to the right (unless they're left-handed). To me, I can get away with anything except the drums being backwards, because it just strikes me funny. However, I thrash a bit at piano, so I always put the low end on the left-hand side and the high end

on the right-hand side.

Hard left and hard right?
Usually, but not always. With a piano, it depends on how phase coherent it was recorded. If it's not dramatic stereo, I'll try to make it more dramatic. Also, if whoever recorded it didn't pay really good attention to the phasing on the mics and the thing is way wide and it falls apart in mono, I'll be panning it in so that in mono it doesn't go away. Sometimes flipping the phase on one side can fix that because a lot of people don't check. Of course, stereo is more important now than ever before, but on a lot of the video channels, you still might be listening in mono, so I check for that.

I try to make stereo records, and I'm not afraid to pan something extremely wide. I like my mixes to have a few things that stick out and get some attention and not just blend in with the crowd, so I always put the electric guitar on the left and steel on the right. That way, there can be all kinds of contrast—not only volume dynamics, but panning dynamics as well.

One of the things I don't like is what I call "big mono," where there's no difference in the left and the right other than a little warble. If you pan that wide left and right, and then here comes another keyboard and you pan that left and right wide, and then there's the two guitars and you pan them left and right wide, by the time you get all this stuff left and right wide, there's really no stereo in the sound. It's like having a big mono record, and it's just not really aurally gratifying. To me, it's better to have some segregation, and that's one of the ways I try to make everything heard in the mixes. Give everybody a place on the stage.

Do you use compression as an effect, or just to even things out, or both?
Both. To me, the key to compression is that it makes the instrument sound like it's being turned up, not being turned down. If you choose the wrong compressor or you use it the wrong way, then your stuff can sound like it's always going away from you. If you use the correct compressor for the job, you can make it sound like, "Man, these guys are coming at you." It's very active and aggressive. Quite often, I'll use it on the stereo buss, but I try not to be too crazy with it.

If you remove all dynamics or if you really lean on it in an improper way during mixing, when it goes to mastering there's not much for the guy to do there. If he does, it'll only compound the problem; then by the time it gets on the radio there's nothing left that'll pump the radio compressors, so then it just kind of lies there. It's loud, but nothing ever really jumps out of your mix.

But yeah, lead vocals almost always get compressed. Bass, certainly when I'm tracking it and quite often when I'm mixing it. I time the release to the tempo of the song or to the end of the note release, especially if the guy's using flat wound strings for more of a retro bass that has a lot of attack and less hang time. Sometimes if you use the wrong compressor on a snare drum, all you'll get is the initial hit and then it'll turn it down, but if you use the right kind of compression, slow

the attack down, speed up the release, you'll get a different effect where there's more length to the snare sound. It'll come sucking back at you. Compression's important, but it's gotta be the right kind, and I think that's the key.

Are you compressing more because of the current tastes of the business?
Country records are a lot more aggressive sounding these days, and that comes from the compression. I don't know if I'm compressing more, but I'm compressing smarter. Rather than piling it all onto one master compressor and flat-topping it to get the volume there, I'm trying to do smarter compression along the way. It can be just as loud and maybe even more exciting, but it doesn't just sound like the guy in the rocket sled with his face pinned back.

I'm using less compression on individual instruments but more on the vocal. Usually low ratio compression where it packs it up gently and you say, "Man, that guy can really work the mic." I'm also doing more intelligent compression on the mix buss.
Ultimately, when you send something out to a client, if you don't crank the level up to at least a level close to what he's used to hearing, you'll lose the gig. When it comes time to deliver it to mastering, I'll drop the level back and leave the mastering engineer some room to work with while retaining the sound that the client has learned to love.

What are you using on the mix buss?
An [Waves] L2 is good. It's kind of gain with no vibe, while the [Waves] L3 is gain with a vibe. The Slate FX-G is gain with vibe and color, where you can let different things poke through using its controls.

Do you add effects as you go along, or do you get the balance up and then add the effects?
I kind of come and go with this. What I'll do is try to make things sound as good as I can dry. If I hear something that just sounds too iso'ed [isolated] and too unrelated to the mix, then I'll add some effects as I go, but I don't go too crazy with it until I get the whole picture. Once it's all sitting there, you can easily tell if it's just not gluing together. My general setup for a mix is I'll have one send set up for long verb and another set up for a short, kind of a room simulation.

Long being what, 2.5, 3 seconds?
Yeah, 2.5, 2.3. For a ballad, sometimes 2.6. Then I'll usually have a delay send with eighth note or sixteenth note or dotted eighth triplets. Sometimes I'll have a little pitch change going. I may have a gated reverb or something that can kind of pull sounds together. Sometimes an isolated guitar sounds great dry and in your face by itself, but other times it seems like, "Wow, they had an overdub party, and look who showed up." Sometimes a little of that gated reverb can kind of smear it together and make it sound like he was actually on the floor with them.

Aren't most country mixes still on the dry side?
Things are still dry, but I worked with a famous country artist who said, "I miss reverb." [Laughs] I'm more into contrast now, where the record may be pretty dry, but you get this long United

Western chamber from Altiverb (the same one that Sinatra and the Beach Boys used) wash on some things. It's a sound, and it puts you in a place.

Some things aren't meant to be all dry, and some things are. Sometimes you just use enough that when you take it away, you go, "Oh, what just happened?" but you don't really notice it otherwise. It'll come full circle one of these days when it will be cool to be wet again.

Are there any go-to plugins that you always use?
As far as effects, one of my favorites is the Reverb 1 [the native Pro Tools plugin], and another I use a lot is the [Audio Ease] Altiverb. [Trillium Lane Labs] TL Space is another that I like, which is similar to the Altiverb. As far as compression, I like the CB 303 by McDSP, which is brilliant. For EQs I like the Massenburg DesignWorks five-band EQ and the seven-band EQ3 by Avid. The thing I like about that is the meter is very accurate. If you don't see red, it's not distorting, unlike some other plugins. It's so good that I'll sometimes put it first in line on my master fader just for the metering.

What do you use for monitoring?
I don't change monitors very often, but for my bigs I now have a Carl Tatz Design PhantomFocus System, which is great. What I love is that I can tell what I've got from DC to radar, and the imaging is so accurate.

I love sitting in front of Carl's monitors—or any large monitors, really—but that's not always a real-world situation. At least 60 percent of the time, I'm off on the side listening to a pair of passive Von Schweikert monitors, which are a little bigger than NS10s. They're sitting off to the side because it's so obvious to hear the balances when you're not in the sweet spot sometimes. They're great for hearing balance and tonality more like the average listener is going to hear it. I still have a boom box that I check every now and then.

Of course, it still amazes me that with all these great and expensive tools, I can still throw it in the car, and in like two seconds I immediately know what's wrong. What I like to do is make it sound good on all three unrelated systems; then it's going to relate to the rest of the world.

How loud do you usually listen when you're mixing?
I mix at different levels. I try not to mix too loud because it'll wear you down and fool your perspective. Sometimes it's very valuable to turn things down, but there's an up and down side to it. If you listen too soft, you'll add too much bass. If you listen too loud, you'll turn the lead vocals down too much.

How many versions of a mix do you do?
People live with a song for so long these days that by the time it gets to the end they pretty much know how it should be. As a result, I'll print a mix and maybe a vocal-up version, but I don't do too many versions since I can get one out almost instantly if someone wants it. Occasionally

someone will ask for a version with less reverb or something like that, but alternate mixes are almost a non-issue anymore.

Are there any tricks you've developed to give you a console sound?
I've been a fan of the Crane Song Phoenix plugin for a long time, and as of late I like the Slate Audio Virtual Console Collection, which gives you different console sounds. It kind of pulls things together in an analog way. Is it exactly the same as a console? No, but it's not better or worse; it's just different. When it comes right down to it, it's still the driver more than the car in most cases.

Chapter 34

Allen Sides

Although well-known as the former owner of the premier Ocean Way Studio complex in Los Angeles, Allen is also one of the most respected engineers in the business, with credits that include film scores and records by the Goo Goo Dolls, Alanis Morissette, Brian Setzer Orchestra, Phil Collins, Natalie Cole, Trisha Yearwood, Wynonna Judd, Kenny Loggins, Michael McDonald, and Aretha Franklin.

Even though he remains on the cutting edge of the latest that recording technology has to offer, Allen continually finds modern uses for many long-forgotten audio relics, proving that sound technique, good ears, and an interesting piece of gear never go out of fashion. Find out about Allen's famed speaker systems at oceanwayaudio.com.

Can you hear the finished mix in your head before you begin?

It depends. I would say that if it's a project that I've been working on, I've already put it up dozens of times, so I have a pretty good idea of what I'm doing. If it's something I'm mixing for someone else, then I listen to their roughs and get a concept of what they have in mind. I really want to understand what they want so I can make that a part of the picture that I draw.

Do you have a special approach to mixing or a philosophy about what you're trying to accomplish?

First, I like it to be fun to listen to. I'll do whatever it takes to make it satisfying. I tend to like a little more lows or extreme highs and a lot of definition, and I like it to sound as punchy as I can make it. So much of that involves the arrangement. When the arrangement is great, then the music comes together in a very nice way. If it fights, then it becomes very difficult to fit together. Getting the arrangement right is an art in itself.

How do you go about building a mix? Where do you normally start from?

I would say that it really varies. Sometimes I'll throw up everything and then after I hear how the vocal sits, I'll look at a section and refine it. Before I do, it's really nice to hear how it relates to the vocals; because you can spend time making the whole thing sound great, but it might not relate to the vocal in any way. I'd say that I listen to the whole thing, then go back and work on each section separately, then put it all together.

Do you have a method for setting levels?

Yeah. When I set up my track, I'll set the monitor level to where I'm comfortable, and I will make it sound as impressive as I can at maybe –2 on the VU meters, because I know I'm going to come up from there. I want to make it as impressive as I possibly can at a fairly modest setting.

This is the whole mix now.

Yeah. I get it to where it's really kicking; then I do my vocal track and get it all happening. Even

when I do that, I probably will end up trimming the individual faders here and there. The problem, of course, is that when you trim the individual faders, the way that they drive the individual effects changes slightly. All the plates and effects sound different when they're driven differently. That's why I try to get everything happening in that lower level, so I have to do as little trimming as possible. I also like to keep my buss masters all the way up. This, of course, depends on the console.

So you're putting the whole mix up first, and then you're adding the vocals later.
Yeah, but as I say, I will probably put the whole mix up, put the vocals in, and listen to how it all fits together before anything. Based on that, I think it's a decision of how I'm going to make the rhythm section sound.

And another thing: I'd say is that I'm definitely a fan of your first impression being your best impression. I like to move very quickly so no matter how complex it is, within two to three hours it's kind of where it should be. A lot of times the music is so complex that you can't actually hear the mix until you put all the mutes in with all the parts playing in the right place. If you just put all the faders up, then you'd have one big mess, so there's a tremendous amount of busy work just to get it prepped so that you can play it back.

Do you have an approach to the rhythm section in particular?
Believe it or not, I typically bring in the overheads first because my overheads are overall drum mics. I bring them in at a good level, then I fit the snare into that, then I get the kick happening. At that point I take a look at the toms and see if they're really adding a lot of good stuff to the drum sound. I'll just keep them on and set them where I want and then push them for fills. If they tend to muddy things up, then I'll set them so they're only on for the tom fills. Obviously you can set certain ambiance and effects on the toms that you don't want on the rest of the kit, and you can make them as big as you want that way. I hate gates. I'd much rather control every fill myself. So it's usually overheads first, then snare, then kick, and then the toms; see how it fits; then tuck in the hat.

Do you have an approach to EQ?
What I would say is that I tend to like things to sound sort of natural, but I don't care what it takes to make it sound like that. Some people get a very preconceived set of notions that you can't do this or you can't do that. Like Bruce Swedien said to me, he doesn't care if you have to turn the knob around backwards; if it sounds good, it is good, assuming that you have a reference point that you can trust, of course.

Do you add effects as you go along, or do you put the mix up and balance it and then add the effects?
No, the effects are usually added as I go along because a lot of times I'll work on multiple image effects on kicks and snares and stuff and tie that into the overheads so you can hear all the sounds as a single entity. Obviously that can change again when the vocals come in. Invariably, what

works by itself is not going to be exactly the same when you put the vocals in. You may have to increase or decrease those effects to get your overall picture to happen.

The other important thing is that when I'm using effects, I hate it to sound generic. I'd much prefer it almost to sound like we're going for a room sound. You have a great natural kick and snare, plenty of attack and punch, and the ambiance surrounds it in such a way that it doesn't sound like an absolute tin-can cheese-ball effect, but becomes more of a natural sound. Obviously, it's relative to the music you're doing.

Are you trying to get something that's more of an ambient sound?
Yeah, there's also a question of dryness versus liveness versus deadness with regard to monitor volume. When you turn it down, your ambiance determines how loud it sounds to you to some degree. If you're monitoring at a loud level and it's very dry, it can be very impressive sounding. When you turn down, it might not be quite so full- sounding, so obviously there's a balance there.

Do you have an approach to panning?
Yeah, I tend to do a lot of hard panning. [Laughs] I don't pan in much since I am really big on having things wide. There's tremendous value in returning to mono, particularly in reverb returns. I still do a lot of comparing between mono and stereo. No matter what anybody says, if you're in a bar, you're going to hear one speaker. There still has to be a relevance between the stereo and mono thing.

How much compression do you use?
Sometimes I use our Focusrite [console] setup, which has three different stereo busses that can combine, and take a mult of the initial totally clean program, and nail it to the wall to bring up all the little ambient stuff, and just tuck that back into the main clean buss so that you can add this sustain that everybody wants without killing the attack. If I take one of my SSL limiters and do that thing that it does, of course it always suffers from a certain lack of impact, so a lot of times we want to get that sustain, particularly on a rocking track, but still want a hell of a punch. That's a way to do that.

Do you have a monitor setup that you usually use, and what level do you listen back at?
I must admit that I really do enjoy our big speakers. I like to turn it up and have fun. I have no problem mixing on anything else, but I like having nice, accurate, big speakers that are fun to listen to that aren't harsh and that don't hurt my ears.

Generally speaking, when I put up the mix, I'll put it up at a fairly good level, maybe 105, and set all my track levels and get it punchy and fun-sounding; then I will probably reference on NS10s at a very modest level, just to check my balance, and go back and forth from there. The small speakers that I'm fond of now, the Genelecs 1032s, I can mix on totally without a problem, but I love my big speakers and I have so much fun. [Laughs]

If I listen loud, it's only for very short periods of time. It's rare that I would ever play a track from beginning to end that loud. I might listen to 20 seconds or 30 seconds of it here and there, but when I'm actually down to really detailing the balance, I'll monitor at a very modest level.

Modest meaning how loud?
I would say that at a level that we could have a conversation and you could hear every word I said.

Do you have any listening tricks, like going to the back of the room or outside and listening through the door?
Oh yeah, I think we all have that. You walk out to get a cup of coffee, and you're listening to it outside in the hall, and the realization strikes you, "How could I miss that?" because it's a different perspective. What I love is my car. I put it on in the car on the way home, and I just call in any changes that I might have, and the assistants print the updates.

How many versions of a mix do you usually do?
Plenty. Invariably I will do the vocal mix to where I'm totally happy with it, and then I'll probably do a quarter and half dB up and a quarter and half dB down. I'll print as many mixes as needed, depending on how difficult the artist is to please. Then if I need to, I'll chop in just the part I want. If there's a word or two, I'll just chop those words in. I really cover myself on mixes these days. I just do not want to have to do a mix again.

Chapter 35

Don Smith

Just one look at producer/engineer/mixer Don Smith's client list gives you an indication of his stature in the music industry. With credits that read like a who's-who of rock and roll, Don has lent his unique expertise to projects by the Rolling Stones, Tom Petty, U2, Stevie Nicks, Bob Dylan, Talking Heads, Eurythmics, the Traveling Wilburys, Roy Orbison, Iggy Pop, the Pointer Sisters, and Bonnie Raitt (and many more). Sadly, Don passed away in 2010 and was warmly remembered in a memorial attended by more than 300 people, including a video from Keith Richards. His method of working was so unique that I wanted to continue to include it even though Don spoke about working back in the console era.

Can you hear the mix before you start?

I can usually hear roughly what it should be. I start out with the basics of a good rough mix and then I try to tweak it from there. Sometimes, I may hear something while I'm doing it, like a tape delay on the drums, that might change the character of the mix and make it turn into a different direction.

How do you start your mix?

Most of the time just drums and bass and then everything else. Then there were some records that I started with lead vocal, then guitar, and the drums would be last. With somebody like Tom Petty, his vocal is so important in the mix that you have to start with the vocal. So the vocals get roughed in, and you throw guitars around it. Then I might start back in the other direction, making sure that the drums and the foundation are solid. But I like to start with the vocal and guitar because it tells me what the song is about and what it's saying; then let everyone else support the song.

Do you have a method for setting levels?

Yeah, I'll start out with the kick and bass in the –7 VU area. By the time you put everything else in, it's +3 (VU) anyway. At least if you start that low, you have room to go higher.

Do you have an approach to using EQ?

Yeah, I use EQ different from some people. I don't just use it to brighten or fatten something up; I use it to make an instrument feel better. Like on a guitar by making sure that all the strings on a guitar can be heard. Instead of just brightening up the high strings and adding mud to the low strings, I may look for a certain chord to hear more of the A string. If the D string is missing in a chord, I like to EQ and boost it way up to +8 or +10 and then just dial through the different frequencies until I hear what they're doing to the guitar. I'm trying to make things more balanced in the way they lay with other instruments.

Do you have a special approach to a lead instrument or vocals?

For vocals, just make sure that the song gets across. The singer is telling a story. He's gotta come through but not be so loud that it sounds like a Pepsi commercial. Sometimes you might want the

vocal to sit back in the track more because it might make the listener listen closer. Sometimes you don't want to understand every word. It depends on the song. It's always different.

Do you build a mix up with effects as you go along?
I always build it up dry. I look at it like building a house. You've got to build a solid foundation first before you can put the decorations on. The same way with tracking. I very rarely use effects when I track. Just every now and again if an effect is an integral part of the track to begin with, then I'll record that.

What I've found is that if you really get it good naked, then when you dress it up, all it can do is get better. If you put on effects too early, then you might disguise something that's not right. I don't really have too many rules about it; I'll just do what feels good at that moment. Sometimes you get it naked and you don't need to put any effects on. It's pretty cool, so just leave it alone.

Do you have a method for adding effects?
I usually start with the delays in time, whether it's eighth note or quarter note or dotted value or whatever. Sometimes on the drums I'll use delays very subtly. If you can hear them, then they're too loud; but if you turn them off, you definitely know they're gone. It adds a natural slap like in a room, so to speak, that maybe you won't hear but you feel.

And, if the drums are dragging, you can speed the delays up just a nat so the drums feel like they're getting a lift. If they're rushing, you can do it the other way by slowing the delays so it feels like they're pulling the track back a bit.

A lot of times in my mixes you won't hear those kinds of things because they're hidden. On the Stones' *Voodoo Lounge* album, there's a song called "Out of Tears." There are these big piano chords that I wanted to sound not so macho and grand, so I put some Phil Spector kind of 15 IPS tape slap on it. It sounded kinda cool, so I tried some on the drums, and it sounded pretty cool there, too. By the end of it, I had it on everything, and it changed the whole song around from a big, grandiose ballad to something more intimate. It was played on a Boesendofer [grand piano], but we really wanted more of an upright, like a John Lennon "Imagine" type of sound.

Do you use tape slap a lot?
I use tape slap all the time. I use it more than I use digital delays. It's a lot warmer and much more natural, and the top end doesn't get so bright and harsh, so it blends in better. I varispeed it to the tempo or whatever feels right. I usually use a four-track with varispeed and an old mono Ampex 440 machine for vocals. The mono has a whole different sound from anything else. Sort of like the Elvis or Jerry Lee Lewis slap, where it can be really loud but never gets in the way because it's always duller yet fatter.

On the four-track, I'll use two channels for stereo, like for drums, and send each slap to the opposite side; then the other tracks I might use for guitars or pre-delay to a chamber or something.

Do you have an approach to panning?
Yeah, it's kinda weird, though. I check my panning in mono with one speaker, believe it or not. When you pan around in mono, all of a sudden you'll find that it's coming through now and you've found the space for it. If I want to find a place for the hi-hat, for instance, sometimes I'll go to mono and pan it around, and you'll find that it's really present all of a sudden, and that's the spot. When you start to pan around on all your drum mics in mono, you'll hear all the phase come together. When you go to stereo, it makes things a lot better.

What level do you listen at?
I like to listen loud on the big speakers to get started, and occasionally thereafter, and most of the time I'll listen at about 90dB. When the mix starts to come together, it comes way down, sometimes barely audible. I turn it down way low and walk around the room to hear everything.

I mix a lot at my house, where I can sit outside on my patio and listen. If I mix in a studio with a lounge, I'll go in there with the control-room door shut and listen like that. I definitely get away from the middle of the speakers as much as possible.

How much compression do you use?
I use a lot of it. Generally, the stereo buss itself will go through a Fairchild 670. Sometimes I'll use a Neve 33609 depending on the song. I don't use much, only a dB or two. There's no rule about it. I'll start with it just on with no threshold, just to hear it.
I may go 20:1 on an 1176 with 20dB of compression on a guitar part as an effect. In general, if it's well recorded, I'll do it just lightly for peaks here and there. I'll experiment with three or four compressors on a vocal. I've got a mono Fairchild to Neves to maybe even a dbx 160 with 10dB of compression to make the vocal just punch through the track.

Again, I don't have any rules. As soon as I think I've got it figured out, on the next song or the next artist, it won't work as well or at all.

Chapter 36

Ed Stasium

Producer/engineer Ed Stasium has made some great guitar albums, such as ones by the Ramones, the Smithereens, and Living Color, but also with the likes of Mick Jagger, Talking Heads, Soul Asylum, Motorhead, and even Gladys Knight and the Pips. In this updated interview, Ed describes how his workflow has changed since switching completely to mixing in the box.

How much is the way you work different these days?

It's nothing like the way I worked before. I no longer work in large facilities, so I do all my mixing in the box, and it works very well. The only thing that's the same is my ears.

Now I'm able to work on many projects simultaneously, and for the most part, I don't have to worry about documentation except for a few pieces of outboard gear, so that makes any recalls fast and easy. When I was using a large- format console, a recall would take hours because I used to use a lot of outboard gear, so working in the box and having a mix come back quickly is great.

The other thing is that I sometimes get an entire session on a flash drive now. On some projects there used to be as many as 60 reels of tape, and now you can fit all that on a 64-gig flash drive.

Mixing is different, too. Now I'll do my preliminary mix and send it off for the artist to hear, and I'll get notes back or have a quick Skype chat. I'll do a revision and send it again. There are usually two or three revisions before we're finished.

Are you using a work surface?

I use a [Avid] Command|8, but I only use one fader at a time. That's the way I even worked on the large-format consoles. I never put all five of my fingers on faders; I always used groups and subgroups for that. I do use the trackball a lot.

Has your mixing philosophy changed after moving completely to the box?

I don't believe that my philosophy is any different. I still try to make things sound the way I think they should sound, which is what I've always done, only now I'm using plugins to do it. I've A/B'd the sound of the stuff that I've done in the past on analog consoles, and my latest stuff sounds pretty good. I actually think that I'm doing the best work I've ever done.

One thing that really helped is that instead of exporting my mixes using Bounce to Disk in Pro Tools, I'm now using a Dangerous Music summing buss, and the difference is remarkable. What I do is use the analog outputs of the 192 [Pro Tools I/O] and go into the Dangerous 2 Buss and take the outputs of that into a dbx 160SL compressor and then into a pair of Rupert Neve Designs 5033 Portico equalizers to do a little EQ on it, then I record it back into Pro Tools. There's a program called Teaboy [by Teaboyaudio.com] that I use for recalls of my outboard gear if I ever need to.

The box makes life easier in a lot of ways, but then it makes life more complicated because you

can do so much more. When I mixed “Midnight Train to Georgia” [the big hit by Gladys Knight and the Pips] back when I first started, we did that on a little 16-input, 16-track console in a basement studio in New Jersey.

I remember the tracks were really packed on that song, so I just brought things in gradually. We started off with the piano, added the guitar, and then added the Hammond. Now I’m riding every snare drum hit to make sure it cuts through, every little guitar nuance, little cymbal things, the kick in certain places, especially the vocal, and a lot of little different effect changes.

“Midnight Train” sounds so clean.
That was a great console, a Langevin. I’m sure the drums were only on one track or two tracks at most. The Pips were double-tracked. You know, Gladys is right up in the front.

We didn’t use many effects on that because we didn’t have any effects. It was a little basement studio, and all we had was a live chamber that was the size of a closet that was concrete with a speaker in there and a couple of microphones. That was the reverb on that record.

Same thing at the Power Station [now called Avatar]. I was mixing the third Ramones record, [*Rocket to Russia*], which was actually the first project mixed there, while we were still building the place. We had a 910 Harmonizer, a couple of Kepexes [the first noise gates], and no reverb at all. What we used for reverb on that whole record was the stairwell.

What go-to plugins do you use?
One of my favorite plugins that I use all the time is the Metric Halo Channel Strip. It reminds me of an SSL in some ways. I also use the Bomb Factory stuff a lot, like their LA-3A and 1176, the Focusrite Scarlett, and the McDSP plug- ins, and I’ll use some of the Joe Meek stuff occasionally. I also like Smack [the stock Pro Tools plugin] on the snare drum, and the EQP-1A I use a lot.

For effects I enjoy the Avid ReVibe, [Audio Ease] TL Space, and the stock Pro Tools D-Verb, which really works great for old-school reverb sounds. And I really like Echo Farm.

Do you have a specific approach when you sit down to mix?
Unlike some other people who are specifically mixers, I’ve been fortunate that everything I produce I’ve been able to follow all the way through to the mix. I’m a “hands on” kind of producer/engineer guy.

Where do you start to build your mix from?
I put the vocals up first and then bring in the bass and drums. I bring up the whole kit at the same time and tweak it. I’m not one to work on the kick drum sound for two hours.

Do you have an approach to panning?
My mixes are kind of mono, but not really. I pan toms but not to extremes, usually between 10 and

2 o'clock. Usually I have the drums in the middle, vocals in the middle, and the solos in the middle. I do pan out the guitars, though. If there's one guitar player, I'll do a lot of double-tracking and have those split out on the sides. If there are two guitar players, I'll just have one guy over on the left and one guy on the right. If there is double-tracking on any of those, I'll split them a little bit, but I never go really wide with them.

Do you use a lot of compressors?
I think of compression as my friend. What I do a lot is take a snare drum and go through an LA-2, just totally com- press it, and then crank up the output so it's totally distorted and edge it in a little bit behind the actual drum. You don't notice the distortion on the track, but it adds a lot of tone in the snare. Actually, something I've done for the last 20 years is to always split out the kick drum and snare drum and take the second output into a Pultec into a dbx 160VU and into a Drawmer 201 gate; then I pretty much overemphasize the EQ and compression on that track and use it in combination with the original track.

What monitors do you use?
I have a 5.1 system of JBL's LSR 4326s. I never use the NS10s anymore, but I do sometimes use my old JBL L100s. I also listen a lot to these small Advent Powered Partners computer speakers. I do all of my final tweaking on them, but I don't listen to them sitting in the middle of the speakers; I listen with one ear to them to make sure things are right.

How many alternate mixes do you do?
I never do them anymore. When I used to do them, they were never used anyway. It literally takes a minute to do an adjustment these days, so there's no need. If I have to adjust the outboard gear, it takes two minutes.

Do you have any mixing tips or tricks?
I like to use Sound Replacer and put the samples underneath the real drums to give them some punch. I'll immediately record the drum samples after a song is tracked because the sound of the drums changes with each song. A really important thing is to make sure that whatever you're recording sounds good in the first place. You're not going to be able to fix anything that doesn't sound good already.

Find out more about Ed at edstasium.com.

Chapter 37

Bruce Swedien

Perhaps no one else in the studio world can so aptly claim the moniker of "Godfather of Recording" as Bruce Swedien. Universally revered by his peers, Bruce has earned that respect thanks to years of stellar recordings for the cream of the musical crop. His credits could fill a book alone, but legends such as Count Basie, Lionel Hampton, Stan Kenton, Duke Ellington, Woody Herman, Oscar Peterson, Nat "King" Cole, George Benson, Mick Jagger, Paul McCartney, Edgar Winter, and Jackie Wilson are good places to start. Then comes Bruce's Grammy-winning projects, which include Michael Jackson's *Thriller* (the biggest-selling record of all time), *Bad, and Dangerous* and Quincy Jones' *Back on the Block* and *Jook Joint*. As one who has participated in the evolution of modern recording from virtually the beginning, as well as being one of its true innovators, Bruce is able to give insights on mixing from a perspective that few of us will ever have.

Do you have a philosophy about mixing that you follow?

The only thing I could say about that is everything that I do in music—mixing or recording or producing—is music driven. It comes from my early days in the studio with Duke Ellington and from there to Quincy. I think the key word in that philosophy is what I would prefer to call "responsibility." From Quincy—no one has influenced me more strongly than Quincy—I've learned that when we go into the studio, our first thought should be that our responsibility is to the musical statement that we're going to make and to the individuals involved, and I guess that's really the philosophy I follow.

Responsibility in that you want to present the music in its best light?

To do it the best way that I possibly can. To use everything at my disposal to not necessarily re-create an unaltered acoustic event, but to present either my concept of the music or the artist's concept of the music in the best way that I can.

Is your concept ever opposed to the artist's concept?

It's funny, but I don't ever remember running into a situation where there's been a conflict. Maybe my concept of the sonics of the music might differ at first with the artist, but I don't ever remember it being a serious conflict.

Is your approach to mixing each song generally the same, then?

No. That's the wonderful part about it. I'll take that a step further and I'll say it's never the same, and I think I have a very unique imagination. I also have another problem in that I hear sounds as colors in my mind [this is actually a neurological condition known as synesthesia]. Frequently when I'm EQing or checking the spectrum of a mix or a piece of music, if I don't see the right colors in it I know the balance is not there.

Can you elaborate on that?

Well, low frequencies appear to my mind's eye as dark colors, black or brown, and high frequencies are brighter colors. Extremely high frequencies are gold and silver. It's funny, but that can be very distracting. It drives me crazy sometimes.

What are you trying to do then, build a rainbow?
No, it's just that if I don't experience those colors when I listen to a mix that I'm working on, I know either that there's an element missing or that the mix values aren't satisfying.

How do you know what proportion of what color should be there?
That's instinctive. Quincy has the same problem. It's terrible! It drives me nuts, but it's not a quantitative thing. It's just that if I focus on a part of the spectrum in a mix and don't see the right colors, it bothers me.

How do you go about getting a balance? Do you have a method?
No, it's purely instinctive. Another thing that I've learned from Quincy, but that started with my work with Duke Ellington, is to do my mixing reactively, not cerebrally. When automated mixing came along, I got really excited because I thought, "At last, here's a way for me to preserve my first instinctive reaction to the music and the mix values that are there." You know how frequently we'll work and work and work on a piece of music, and we think, "Oh boy, this is great. Wouldn't it be great if it had a little more of this or a little more of that?" Then you listen to it in the cold, gray light of dawn, and it sounds pretty bad. Well, that's when the cerebral part of our mind takes over, pushing the reactive part to the background, so the music suffers.

Do you start to do your mix from the very first day of tracking?
Yes, but again I don't think that you can say any of these thoughts is across the board. There are certain types of music that grow in the studio. You go in and start a rhythm track and think you're gonna have one thing, and all of a sudden it does a sharp left and it ends up being something else. There are other types of music where I start the mix even before the musicians come to the studio. I'll give you a good example of something. On Michael's HIStory album, for the song "Smile, Charlie Chaplin," I knew what that mix would be like two weeks before the musicians hit the studio.

From listening to the demo?
No. It had nothing to do with anything except what was going on in my mind, because Jeremy Lubbock, the orchestra arranger and conductor, and I had talked about that piece of music and the orchestra that we were going to use. I came up with a studio setup that I had used with the strings of the Chicago Symphony many years before at Universal, where the first violins are set up to the left of the conductor and the second violins to the right, the violas behind the first fiddles and the celli behind the second fiddles, which is a little unusual, so I had that whole mix firmly in mind long before we did it.

So sometimes you do hear the final mix before you start.

Sometimes, but that's rare.

Where do you generally build your mix from?
It's totally dependent on the music, but if there were a method to my approach, I would say the rhythm section. You usually try to find the motor and then build the car around it.

Some people say they always put the bass up first...some from the snare, some the overheads….
No, I don't think I have any set way. I think it would spoil the music to think about it that much.

Do you have a method for panning?
I don't think I have any approach to it. I generally do whatever works with the music that I'm doing.

So it's just something that hits you when you're doing it?
Yeah, that's really the way it works. It'll be an idea, whether it's panning or a mix value or an effect or whatever, and I'll say, "Ooh, that's great. I'm gonna do that."

What level do you usually monitor at?
That's one area where I think I've relegated it to a science. For the nearfield speakers, I use Westlake BBSM8s, and I try not to exceed 85dB SPL. On the Auratones I try not to exceed 83. What I've found in the past few years is that I use the big speakers less and less with every project.

Are you listening in mono on the Auratones?
Stereo.

Do you listen in mono much?
Once in a while. I always check it because there are some places where mono is still used.

I love the way you sonically layer things when you mix. How do you go about getting that?
I have no idea. If knew, I probably couldn't do it as well. It's purely reactive and instinctive. I don't have a plan. Actually, what I will do frequently when we're layering with synths is to add some acoustics to the synth sounds. I think this helps in the layering in that the virtual direct sound of most synthesizers is not too interesting, so I'll send the sound out to the studio and use a coincident pair of mics to blend a little bit of acoustics back with the direct sound. Of course it adds early reflections to the sound, which reverb devices can't do. That's the space before the onset of reverb where those early reflections occur.

So what you're looking for more than anything is early reflections?
I think that's a much overlooked part of sound because there are no reverb devices that can generate that. It's very important. Early reflections will usually occur under 40 milliseconds. It's a fascinating part of sound.

When you're adding effects, are you using mostly reverbs or delays?
A combination. Lately, though, I have been kinda going through a phase of using less reverb. I've got two seven- foot-high racks full of everything. I have an EMT 250, a 252, and all the usual stuff, all of which I bought new. No one else has ever used them. It's all in pretty good shape, too.

Do you have any listening tricks?
Since I moved from California [Bruce now lives in Ocala, Florida], one of the things that I miss is my time in the car. I had a Ford Bronco with an incredible sound system, and I still kinda miss that great listening environment.

Do you do all your work at your facility now?
No, I go wherever they'll have me. I love it here, but my studio's dinky. I have an older little 40-input Harrison and a 24-track. The Harrison is a wonderful desk. It's a 32 series and the same as the one I did Thriller on. Actually, I think that's one of the most underrated desks in the industry. It's all spiffied up with a beautiful computer and Neve summing amps. It's just fabulous.

How long does it usually take you to do a mix?
That can vary. I like to try not to do more than one song a day unless it's a really simple project, and then I like to sleep on a mix and keep it on the desk overnight. That's one of the advantages of having my little studio at home.

How many versions of a mix do you do?
Usually one. Although when I did "Billie Jean," I did 91 mixes of that thing, and the mix that we finally ended up using was Mix 2. I had a pile of half-inch tapes to the ceiling. All along we thought, "Oh man, it's getting better and better." [Laughs]

Do you have an approach to using EQ?
I don't think I have a philosophy about it. What I hate to see is an engineer or producer start EQing before they've heard the sound source. To me it's kinda like salting and peppering your food before you've tasted it. I always like to listen to the sound source first, whether it's recorded or live, and see how well it holds up without any EQ or whatever.

That being the case, do you have to approach things differently if you're just coming in to do the mix?
Not usually, but I'm not really crazy about working on things that other people have recorded—I gotta tell you that. I consider myself fortunate to be working, so that's the bottom line. [Laughs]

Do you add effects as you go?
There are probably only two effects that I use on almost everything, and that's the EMT 250 and the 252. I love those reverbs. There's nothing in the industry that comes close to a 250 or a 252.

What are you using the 252 on?
I love the 252 on vocals with the 250 program. It's close to a 250, but it's kinda like a 250 after taxes. It's wonderful, but there's nothing like a 250.

What do you do to make a mix special?
I wish I knew. The best illustration of something special is when we were doing "Billie Jean," and Quincy said, "Okay, this song has to have the most incredible drum sound that anybody has ever done, but it also has to have one element that's different, and that's sonic personality." I lost a lot of sleep over that. What I ended up doing was building a drum platform and designing some special little things like a bass drum cover and a flat piece of wood that goes between the snare and the hat. And the bottom line is that there aren't many pieces of music where you can hear the first three or four notes of the drums and immediately tell what song it is. I think that's the case with "Billie Jean," and that I attribute to sonic personality, but I lost a lot of sleep over that one before it was accomplished.

Do you determine that personality before you start to record?
Not really, but in that case I got to think about the recording setup in advance. And of course, I have quite a microphone collection that goes with me everywhere (17 anvil cases!), and that helps a little bit in that they're not beat up.

Are most of the projects that you do these days both tracking and mixing?
I don't know what's happened, but I don't get called to record stuff very much these days. People are driving me nuts with mixing, and I love it, but I kinda miss tracking. A lot of people think that since I moved to Florida I retired or something, but that's the last thing I'd want to do. You know what Quincy and I say about retiring? Retiring is when you can travel around and get to do what you want. Well, I've been doing that all my life. I love what I do, and I'm just happy to be working, so that's the bottom line.

Glossary

0 dB Full Scale. Abbreviated FS, it's the highest level that can be recorded in the digital domain. Recording beyond 0 dB FS results in severe distortion.

5.1. A speaker system that uses three speakers across the front and two stereo speakers in the rear, along with a subwoofer.

808. One of the early drum machines made by Roland.

airplay. When a song gets played on the radio.

ambience. The background noise of an environment.

arpeggio. The notes of a chord played in quick succession.

arrangement. The way the instruments are combined in a song.

articulations. The way a note or phrase is played or sung in terms of attack and release.

attack. The first part of a sound. On a compressor/limiter, a control that affects how that device will respond to the attack of a sound.

attenuation. A decrease in level.

attenuation pad (sometimes just called a *pad*). A circuit that decreases the input level by a set amount. The amount of attenuation is usually in 10dB or 20dB increments.

automation. A system that memorizes and then plays back the position of all faders and mutes on a console. In a DAW, the automation can also record and play back other parameters, including sends, returns, panning, and plug-ins.

B-section. A section of a song between the verse and chorus sections. Not found in every song.

bandwidth. The number of frequencies that a device will pass before the signal degrades. A human being can supposedly hear from 20Hz to 20kHz, so the bandwidth of the human ear is 20Hz to 2 kHz.

basic track. Recording the rhythm section for a record, which could be only the drums but could also include all the instruments of the band, depending upon the project.

bass management. A circuit that utilizes the subwoofer in a 5.1 system to provide bass extension

for the five main speakers. The bass manager steers all frequencies below 80 Hz into the subwoofer along with the LFE source signal. *See* LFE.

bass redirection. Another term for bass management.

big ears. The ability to be very aware of everything going on within the session and with the music. The ability to rapidly dissect a track in terms of arrangement.

bit rate. The transmission rate of a digital signal.

bottom. Bass frequencies, the lower end of the audio spectrum. *See also* low end.

bottom end. *See* bottom.

bpm. Beats per minute. The measure of tempo.

brick wall. A limiter employing "look-ahead" technology that is so efficient that the signal will never exceed a certain predetermined level, so there can be no digital "overs."

buss. A signal pathway.

butt cut. Sometimes known as a *straight cut*, a butt cut is an audio edit at a 90-degree angle with no fade.

chamber (reverb). A method to create artificial reverberation using a tiled room in which a speaker and several microphones are placed.

chatter. When a gate rapidly turns on and off due to fluctuating signal dynamics.

chorus. A type of signal processor where a detuned copy is mixed with the original signal to create a fatter sound.

clean. A signal with no distortion.

clip. To overload and cause distortion.

clipping. When an audio signal begins to distort because a circuit in the signal path is overloaded, the top of the waveform becomes "clipped" off and begins to look square instead of rounded. This usually results in some type of distortion, which can be either soft and barely noticeable or horribly crunchy-sounding.

codec. An acronym for en**co**der/**dec**oder.

color. To affect the timbral qualities of a sound.

comb filter. A distortion produced by combining an electronic or acoustic signal with a delayed copy of itself. The result is peaks and dips introduced into the frequency response.

compression. Signal processing that controls and evens out the dynamics of a sound.

compressor. A signal-processing device used to compress audio dynamics.

competitive level. A mix level that is as loud as your competitor's mix.

cut. To decrease, attenuate, or make less.

DAC. Digital-to-analog convertor. The device that converts the signal from the digital domain to the analog domain.

data compression. An algorithm that selectively eliminates bits from a digital stream to make it more efficient for storage and transmission.

DAW. A digital audio workstation. A computer with the appropriate hardware and software needed to digitize and edit audio.

dB. Stands for *decibel*, which is a unit of measurement of sound level or loudness. The smallest change in level that an average human can hear is 1 dB, according to many textbooks.

decay. The time it takes for a signal to fall below audibility.

delay. A type of signal processor that produces distinct repeats (echoes) of a signal.

desk. A British name for a recording console.

DI. Direct inject, an impedance-matching device for guitar or bass that eliminates the need for a microphone.

direct. To "go direct" means to bypass a microphone and connect the guitar, bass, or keyboard directly into a recording device.

direct box. *See* DI.

digital domain. When a signal source is converted into a series of electronic pulses represented by 1s and 0s, the signal is then in the digital domain.

digital over. The point beyond 0 on a digital processor where the red Over indicator lights, resulting in a digital overload and distortion.

divergence. A parameter of surround panning that allows you to increase the level to channels other than the one panned to.

double. To play or sing a track a second time. The inconsistencies between both tracks make the part sound bigger.

downmix. When a multichannel surround mix is electronically interpreted into stereo.

dynamics. The volume execution when an instrument is played. Songs that vary in dynamics are found to be expressive and interesting.

dynamic range. A ratio that describes the difference between the loudest and the quietest audio. The higher the number, equaling a greater dynamic range, the better.

dubbing mixer. A film mixer who works on a dubbing stage, which is a film theater with a console placed in the middle.

edgy. A sound with an abundance of midrange frequencies.

element. A component or ingredient of the mix.

envelope. The attack, sustain, and release of a sound.

equalizer. A tone control that can vary in sophistication from very simple to very complex. *See* parametric equalizer.

equalization. Adjustment of the frequency spectrum to even out or alter tonal imbalances.

exciter. An outboard effects device that uses phase manipulation and harmonic distortion to produce high-frequency enhancement of a signal.

feedback. When part of the output signal is fed back into the input.

feel. The groove of a song and how it feels to play or listen to it.

flanging. The process of mixing a copy of the signal back with itself, but gradually and randomly slowing the copy down, causing the sound to “whoosh” as if it were in a wind tunnel. This was originally done by holding a finger against a tape flange (the metal part that holds the magnetic

tape on the reel), hence the name.

flip the phase. Selecting the phase switch on a console, preamp, or DAW channel in order to find the setting with the greatest bass response.

footballs. Musical whole notes. Long sustaining chords.

FS. Full scale. A digital peak meter that reads at dB shows the full scale of the meter. The maximum amplitude of a digital system.

gain. The amount that a sound is boosted.

gain reduction. The amount of compression or limiting.

gain staging. Setting the gain of each stage in the signal path so that one stage doesn't overload the next one in line.

grid. The spaced lines on a DAW timeline that represents each beat and sub-beat.

groove. The pulse of the song and how the instruments dynamically breathe with it. Or, the part of a vinyl record that contains the mechanical information that is transferred to electronic info by the stylus.

Haas Effect. A psychoacoustic effect where any delay signal below 40 milliseconds is indistinguishable from the source event. In other words, instead of hearing the sound and then a delay (two events), you hear both the source and the delay together as a single event.

headroom. The amount of dynamic range between the normal operating level and the maximum level, which is usually the onset of clipping.

Hz. An abbreviation for hertz, which is the measurement unit of audio frequency, meaning the number of cycles per second. High numbers represent high sounds, and low numbers represent low sounds.

high end. The high frequency response of a device.

high-pass filter. An electronic device that allows the high frequencies to pass while attenuating the low frequencies. Used to eliminate low-frequency artifacts like hum and rumble. The frequency point where it cuts off can be fixed, switchable, or variable.

hook. A catchy phrase either played or sung.

hyper-compression. Too much buss compression during mixing or limiting during mastering in an effort to make the recording louder results in what's known as *hyper-compression*, a condition that essentially leaves no dynamics and makes the track sound lifeless.
I/O. The input/output of a device.

input pad. An electronic circuit that attenuates the signal, usually 10 or 20dB. *See also* attenuation pad.

in the box. Doing all of your mixing with the software console in the DAW application on the computer, instead of using a hardware console.

iso booth. Isolation booth. An isolated section of the studio designed to eliminate leakage from coming into the booth or leaking out.

intonation. The accuracy of tuning anywhere along the neck of a stringed instrument like a guitar or bass. Also applies to brass, woodwinds, and piano.

knee. The speed at which a compressor will turn on once it reaches threshold. A soft knee turns on gradually and is less audible than a hard knee.

kHz. One kHz equals 1,000 hertz (example: 4kHz = 4,000 Hz).

lacquer. The vinyl master, which is a single-sided 14" disc made of aluminum substrate covered with a soft cellulose nitrate. A separate lacquer is required for each side of a vinyl record. Since the lacquer can never be played, a ref or acetate is made to check the disc.

latency. Latency is a measure of the time it takes (in milliseconds) for your audio signal to pass through your system during the recording process. This delay is caused by the time it takes for your computer to receive, understand, process, and send the signal back to your outputs.

leakage. Sound from a distant instrument "bleeding" into a mic pointed at another instrument. Acoustic spill from a sound source other than the one intended for pickup.

Leslie. A speaker cabinet that features rotating speakers primarily used with organs.

LFE. Low-frequency effects channel. This is a special channel of 30Hz to 120Hz information primarily intended for special effects, such as explosions in movies. The LFE has an additional 10dB of headroom to accommodate the required sound pressure level of the low frequencies.

limiter. A signal-processing device used to constrict or reduce audio dynamics, reducing the loudest peaks in volume.

look-ahead. In a digital limiter, look-ahead delays the audio signal a small amount (about two milliseconds) so that the limiter can anticipate the transients in such a way that it catches the peak before it gets by.

loop. A small audio file, usually only four or eight beats (or measures) long that's edited in a way so that it can seamlessly repeat.

low-pass filter (LPF). A electronic frequency filter that allows only the low frequencies to pass while attenuating the high frequencies. The frequency point where it cuts off is usually either switchable or variable.

low end. The lower end of the audio spectrum, or bass frequencies usually below 200Hz.

make-up gain. A control on a compressor/limiter that applies additional gain to the signal. This is helpful because the signal is automatically decreased when the compressor is working. Make-up gain "makes up" the gain and brings it back to where it was prior to being compressed.

master. A final version of a recording that is destined for distribution.

mastering. The process of turning a collection of songs into a record by making them sound like they belong together in tone, volume, and timing (spacing between songs on an album).

metadata. Data that describes the primary data. For instance, metadata can be data about an audio file that indicates the date recorded, sample rate, resolution, and so on.

midrange. Middle frequencies starting from around 250Hz and going up to 4,000Hz.

mix buss. The network that mixes all of the individual channels together for your final mix.

modeling. Developing a software algorithm that is an electronic representation of the sound of a hardware audio device down to the smallest behaviors and nuances.

modulation. Using a second signal to modify the first. For example, a chorus uses a very low-frequency signal to modulate the audio signal and produce the effect.

mono. Short for *monaural*, or single audio playback channel.

monaural. A mix that contains a single channel and usually comes from only one speaker.

MP3. A standard data-compression format used to make audio files smaller in size.

muddy. Non-distinct because of excessive low frequencies.

mult. A section of the patchbay that enables patching to multiple inputs or outputs.

multi-band compression. A compressor that is able to individually compress different frequency bands as a means of providing more control over the compression process.

mute. An on/off switch. To mute something means to turn it off.

outboard gear. Hardware devices such as compressors, reverbs, and effects boxes that are not built into a console and usually reside in an equipment rack in the control room.

outro. The section of a song after the last chorus until the end of the song.

overs. Digital overs occur when the level is so high that it attempts to go beyond 0 dB Full Scale on a typical digital level meter found in just about all digital equipment. A red Overload indicator usually will light, accompanied by the crunchy, distorted sound of waveform clipping.

overdub. To record a new track while listening to previously recorded tracks.

overtone. The part of a sound that gives it its character and uniqueness.

out of phase. The polarity of two channels (it could be the left and right channels of a stereo program) are reversed, thereby causing the center of the program (such as the vocal) to diminish in level. Electronically, when one cable is wired backwards from all the others.

pan. Short for *panorama*. Indicates the position of an instrument within the stereo spectrum.

panning. Moving a sound across the stereo spectrum.

parametric equalizer. A tone control where the gain, frequency, and bandwidth are all variable.

peaks. A sound that's temporarily much higher than the sound surrounding it.

phase. The relationship between two separate sound signals when combined into one.

phantom image. In a stereo system, if the signal is of equal strength in the left and right channels, the resultant sound appears to come from in between them.

phase shift. The process during which some frequencies (usually those below 100Hz) are slowed down ever so slightly as they pass through a device. This is usually exaggerated by excessive use of equalization and is highly undesirable.

phase meter. A dedicated meter that displays the relative phase of a stereo signal.

plate (reverb). A method used to create artificial reverberation using a large steel plate with a speaker and several transducers connected to it.

plug-in. An add-on to a computer application that adds functionality to it. EQ, modulation, and reverb are examples of DAW plug-ins.

point. The frequencies between 2k and 5kHz that cause a sound to be more distinct.

power chords. Long, sustaining, distorted guitar chords.

predelay. A variable length of time before the onset of reverberation. Predelay is often used to separate the source from the reverberation so the source can be heard more clearly.

preroll. A short length of time before recording begins or a song section arrives.

presence. Accentuated upper midrange frequencies (anywhere from 5k to 10kHz).

producer. The equivalent of a movie director, the producer has the ability to craft the songs of an artist or band technically, sonically, and musically.

proximity effect. The inherent low-frequency boost that occurs with a directional microphone as it gets closer to the signal source.

Pultec. An equalizer made during the ’50s and ‘60s by Western Electric that is highly prized today for its smooth sound.

pumping. When the level of a mix increases and then decreases noticeably. Pumping is caused by the improper setting of the attack and release times on a compressor.

punchy. A description for a quality of sound that infers good reproduction of dynamics with a strong impact. The term sometimes means emphasis in the 200Hz and 5kHz areas.

Q. Bandwidth of a filter or equalizer.

range. On a gate or expander, a control that adjusts the amount of attenuation that will occur to the signal when the gate is closed.

ratio. A control on a compressor/limiter that determines how much compression or limiting will occur when the signal exceeds the threshold.

recall. A system that memorizes the position of all pots and switches on a console. The engineer must still physically reset the pots and switches back to their previous positions as indicated on a video monitor.

record. A generic term for the distribution method of a recording. Regardless of whether it's a CD, vinyl, or a digital file, it is still known as a record.

release. The last part of a sound. On a compressor/limiter, a control that affects how that device will respond to the release of a sound.

resonance. *See* resonant frequency.

resonant frequency. A particular frequency or band of frequencies that is accentuated, usually due to some extraneous acoustic, electronic, or mechanical factor.

return. An input on a recording console especially dedicated for effects devices, such as reverbs and delays. The return inputs are usually not as sophisticated as normal channel inputs on a console.

reverb. A type of signal processor that reproduces the spatial sound of an environment (such as the sound of a closet or locker room or inside an oil tanker).

rhythm section. The instruments in a band that give the song its pulse, usually the bass and drums.

RMS meter. A meter that reads the average level of a signal.

roll off. To attenuate either end of the frequency spectrum.

scratch vocal. A temporary vocal recorded during basic tracking with the intention of replacing it later.

shelving curve. A type of equalizer circuit used to boost or cut a signal above or below a specified frequency; looks flat like a shelf when graphed. Usually the high- and low-band equalizers built into many mixing boards are the shelving type.

sibilance. A short burst of high frequencies centering anywhere in a vocal's 3kHz to 10kHz range, resulting in the "S" sounds being overemphasized.

sidechain. A separate signal path to and from the control element of a dynamics device.

signal path. The electronic or digital circuitry that the audio signal must pass through.

soundfield. The listening area containing mostly direct sound from the monitor speakers.

source. An original master that is not a copy or a clone.

spectrum. The complete audible range of audio signals.

SPL. Sound-pressure level. The volume level of a sound to the human ear.

stage. In an analog console, a block of circuitry that performs a console function, such as EQ or panning. In a digital or software console, a digital block that performs a console function.

standing waves. An acoustic property of a room where certain frequencies reflect off the walls, floor, or ceiling that will either boost the signal or attenuate it, depending upon where in the room you're standing.

sympathetic vibration. Vibrations, buzzes, and rattles or notes that occur in areas of an instrument or instruments other than the one that was struck.

subgroup. A separate submixer that sums the assigned channels together and then sends that mix to the master mix buss.

sub. Short for *subwoofer*.

subwoofer. A low-frequency speaker with a frequency response from about 30Hz to as high as 120Hz.

synchronization. When two devices—usually storage devices, such as tape machines, DAWs, or sequencers—are locked together with respect to time.

tape slap. A method to create a delay effect using a tape machine.

tempo. The rate of speed at which a song is played.

tension and release. Building a listener's expectations and then relaxing them, such as dissonance to harmony or loud to soft.

threshold. The point at which an effect takes place. On a compressor/limiter, for instance, the threshold control adjusts the signal level at which compression will take place.

track sharing. When a single track shares more than one instrument. For instance, when a percussion part is recorded on a guitar solo track in places that the guitar has not been recorded.

timed delay. A delay where the repeats are timed to pulse along with the pulse of the song.

top-end. *See* high-end.

track. A term sometimes used to mean a song. In recording, a separate musical performance that is recorded.

transient. A very short-duration signal.

TV mix. A mix without the lead vocals so the artist can sing live to the backing tracks during a television appearance.

vamp. To repeat a short passage of music.

Delay Chart

BPM	1/4 Note	8th Note	16th Note	32nd Note	1/4 Triplet	8th Triplet	16th Triplet	Dotted 1/4	Dotted 8th	Dotted 16th
75	800	400	200	100	533.6	266.8	133.4	1200	600	300
76	789.5	394.7	197.4	98.7	526.6	263.3	131.6	1184.2	592.1	296.1
77	779.2	389.6	194.8	97.4	519.7	259.9	129.9	1168.8	584.4	292.2
78	769.2	384.6	192.3	96.2	513.1	256.5	128.3	1153.8	576.9	288.5
79	759.5	379.7	189.9	94.9	506.6	253.3	126.6	1139.2	569.6	284.8
80	750	375	187.5	93.8	500.3	250.1	125.1	1125	562.5	281.3
81	740.7	370.4	185.2	92.6	494.1	247	123.5	1111.1	555.6	277.8
82	731.7	365.9	182.9	91.5	488	244	122	1097.6	548.8	274.4
83	722.9	361.4	180.7	90.4	482.2	241.1	120.5	1084.3	542.2	271.1
84	714.3	357.1	178.6	89.3	476.4	238.2	119.1	1071.4	535.7	267.9
85	705.9	352.9	176.5	88.2	470.8	235.4	117.7	1058.8	529.4	264.7
86	697.7	348.8	174.4	87.2	465.3	232.7	116.3	1046.5	523.3	261.6
87	689.7	344.8	172.4	86.2	460	230	115	1034.5	517.2	258.6
88	681.8	340.9	170.5	85.2	454.8	227.4	113.7	1022.7	511.4	255.7
89	674.2	337.1	168.5	84.3	449.7	224.8	112.4	1011.2	505.6	252.8
90	666.7	333.3	166.7	83.3	444.7	222.3	111.2	1000	500	250
91	659.3	329.7	164.8	82.4	439.8	219.9	109.9	989	494.5	247.3
92	652.2	326.1	163	81.5	435	217.5	108.8	978.3	489.1	244.6
93	645.2	322.6	161.3	80.6	430.3	215.2	107.6	967.7	483.9	241.9
94	638.3	319.1	159.6	79.8	425.7	212.9	106.4	957.4	478.7	239.4
95	631.6	315.8	157.9	78.9	421.3	210.6	105.3	947.4	473.7	236.8
96	625	312.5	156.3	78.1	416.9	208.4	104.2	937.5	468.8	234.4
97	618.6	309.3	154.6	77.3	412.6	206.3	103.1	927.8	463.9	232
98	612.2	306.1	153.1	76.5	408.4	204.2	102.1	918.4	459.2	229.6
99	606.1	303	151.5	75.8	404.2	202.1	101.1	909.1	454.5	227.3
100	600	300	150	75	400.2	200.1	100.1	900	450	225
101	594.1	297	148.5	74.3	396.2	198.1	99.1	891.1	445.5	222.8
102	588.2	294.1	147.1	73.5	392.4	196.2	98.1	882.4	441.2	220.6
103	582.5	291.3	145.6	72.8	388.5	194.3	97.1	873.8	436.9	218.4
104	576.9	288.5	144.2	72.1	384.8	192.4	96.2	865.4	432.7	216.3
105	571.4	285.7	142.9	71.4	381.1	190.6	95.3	857.1	428.6	214.3
106	566	283	141.5	70.8	377.5	188.8	94.4	849.1	424.5	212.3
107	560.7	280.4	140.2	70.1	374	187	93.5	841.1	420.6	210.3
108	555.6	277.8	138.9	69.4	370.6	185.3	92.6	833.3	416.7	208.3
109	550.5	275.2	137.6	68.8	367.2	183.6	91.8	825.7	412.8	206.4
110	545.5	272.7	136.4	68.2	363.8	181.9	91	818.2	409.1	204.5
111	540.5	270.3	135.1	67.6	360.5	180.3	90.1	810.8	405.4	202.7
112	535.7	267.9	133.9	67	357.3	178.7	89.3	803.6	401.8	200.9
113	531	265.5	132.7	66.4	354.2	177.1	88.5	796.5	398.2	199.1
114	526.3	263.2	131.6	65.8	351.1	175.5	87.8	789.5	394.7	197.4
115	521.7	260.9	130.4	65.2	348	174	87	782.6	391.3	195.7
116	517.2	258.6	129.3	64.7	345	172.5	86.3	775.9	387.9	194
117	512.8	256.4	128.2	64.1	342.1	171	85.5	769.2	384.6	192.3
118	508.5	254.2	127.1	63.6	339.2	169.6	84.8	762.7	381.4	190.7
119	504.2	252.1	126.1	63	336.3	168.2	84.1	756.3	378.2	189.1
BPM	1/4 Note	8th Note	16th Note	32nd Note	1/4 Triplet	8th Triplet	16th Triplet	Dotted 1/4	Dotted 8th	Dotted 16th
120	500	250	125	62.5	333.5	166.8	83.4	750	375	187.5
121	495.9	247.9	124	62	330.7	165.4	82.7	743.8	371.9	186
122	491.8	245.9	123	61.5	328	164	82	737.7	368.9	184.4
123	487.8	243.9	122	61	325.4	162.7	81.3	731.7	365.9	182.9
124	483.9	241.9	121	60.5	322.7	161.4	80.7	725.8	362.9	181.5
125	480	240	120	60	320.2	160.1	80	720	360	180
126	476.2	238.1	119	59.5	317.6	158.8	79.4	714.3	357.1	178.6
127	472.4	236.2	118.1	59.1	315.1	157.6	78.8	708.7	354.3	177.2
128	468.8	234.4	117.2	58.6	312.7	156.3	78.2	703.1	351.6	175.8
129	465.1	232.6	116.3	58.1	310.2	155.1	77.6	697.7	348.8	174.4
130	461.5	230.8	115.4	57.7	307.8	153.9	77	692.3	346.2	173.1
131	458	229	114.5	57.3	305.5	152.7	76.4	687	343.5	171.8
132	454.5	227.3	113.6	56.8	303.2	151.6	75.8	681.8	340.9	170.5
133	451.1	225.6	112.8	56.4	300.9	150.5	75.2	676.7	338.3	169.2
134	447.8	223.9	111.9	56	298.7	149.3	74.7	671.6	335.8	167.9
135	444.4	222.2	111.1	55.6	296.4	148.2	74.1	666.7	333.3	166.7
136	441.2	220.6	110.3	55.1	294.3	147.1	73.6	661.8	330.9	165.4
137	438	219	109.5	54.7	292.1	146.1	73	656.9	328.5	164.2
138	434.8	217.4	108.7	54.3	290	145	72.5	652.2	326.1	163
139	431.7	215.8	107.9	54	287.9	144	72	647.5	323.7	161.9
140	428.6	214.3	107.1	53.6	285.9	142.9	71.5	642.9	321.4	160.7
141	425.5	212.8	106.4	53.2	283.8	141.9	71	638.3	319.1	159.6
142	422.5	211.3	105.6	52.8	281.8	140.9	70.5	633.8	316.9	158.5
143	419.6	209.8	104.9	52.4	279.9	139.9	70	629.4	314.7	157.3
144	416.7	208.3	104.2	52.1	277.9	139	69.5	625	312.5	156.3
145	413.8	206.9	103.4	51.7	276	138	69	620.7	310.3	155.2
146	411	205.5	102.7	51.4	274.1	137.1	68.5	616.4	308.2	154.1

147	408.2	204.1	102	51	272.2	136.1	68.1	612.2	306.1	153.1
148	405.4	202.7	101.4	50.7	270.4	135.2	67.6	608.1	304.1	152
149	402.7	201.3	100.7	50.3	268.6	134.3	67.1	604	302	151
150	400	200	100	50	266.8	133.4	66.7	600	300	150
151	397.4	198.7	99.3	49.7	265	132.5	66.3	596	298	149
152	394.7	197.4	98.7	49.3	263.3	131.6	65.8	592.1	296.1	148
153	392.2	196.1	98	49	261.6	130.8	65.4	588.2	294.1	147.1
154	389.6	194.8	97.4	48.7	259.9	129.9	65	584.4	292.2	146.1
155	387.1	193.5	96.8	48.4	258.2	129.1	64.5	580.6	290.3	145.2
156	384.6	192.3	96.2	48.1	256.5	128.3	64.1	576.9	288.5	144.2
157	382.2	191.1	95.5	47.8	254.9	127.5	63.7	573.2	286.6	143.3
158	379.7	189.9	94.9	47.5	253.3	126.6	63.3	569.6	284.8	142.4
159	377.4	188.7	94.3	47.2	251.7	125.8	62.9	566	283	141.5
160	375	187.5	93.8	46.9	250.1	125.1	62.5	562.5	281.3	140.6

About Bobby Owsinski

Producer/engineer Bobby Owsinski is one of the best selling authors in the music industry with 23 books that are now staples in audio recording, music, and music business programs in colleges around the world, including the *Deconstructed Hits series, Music 4.1 Internet Music Guidebook, The Mixing Engineer's Handbook* and more. He's also a contributor to Forbes writing on the new music business, his popular blogs have passed 7 million visits, and he's appeared on CNN and ABC News as a music branding and audio expert.

Visit Bobby's music production blog at bobbyowsinskiblog.com, his Music 3.0 music industry blog at music3point0.com, his Forbes blog at forbes.com/sites/bobbyowsinski/, his podcast at bobbyoinnercircle.com, and his website at bobbyowsinski.com. Bobby's mixing and production courses can be found at bobbyowsinskicourses.com.

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